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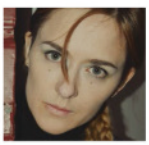
May 2024 / CM333

THE ULTIMATE GUIDE TO CREATIVE REVERB

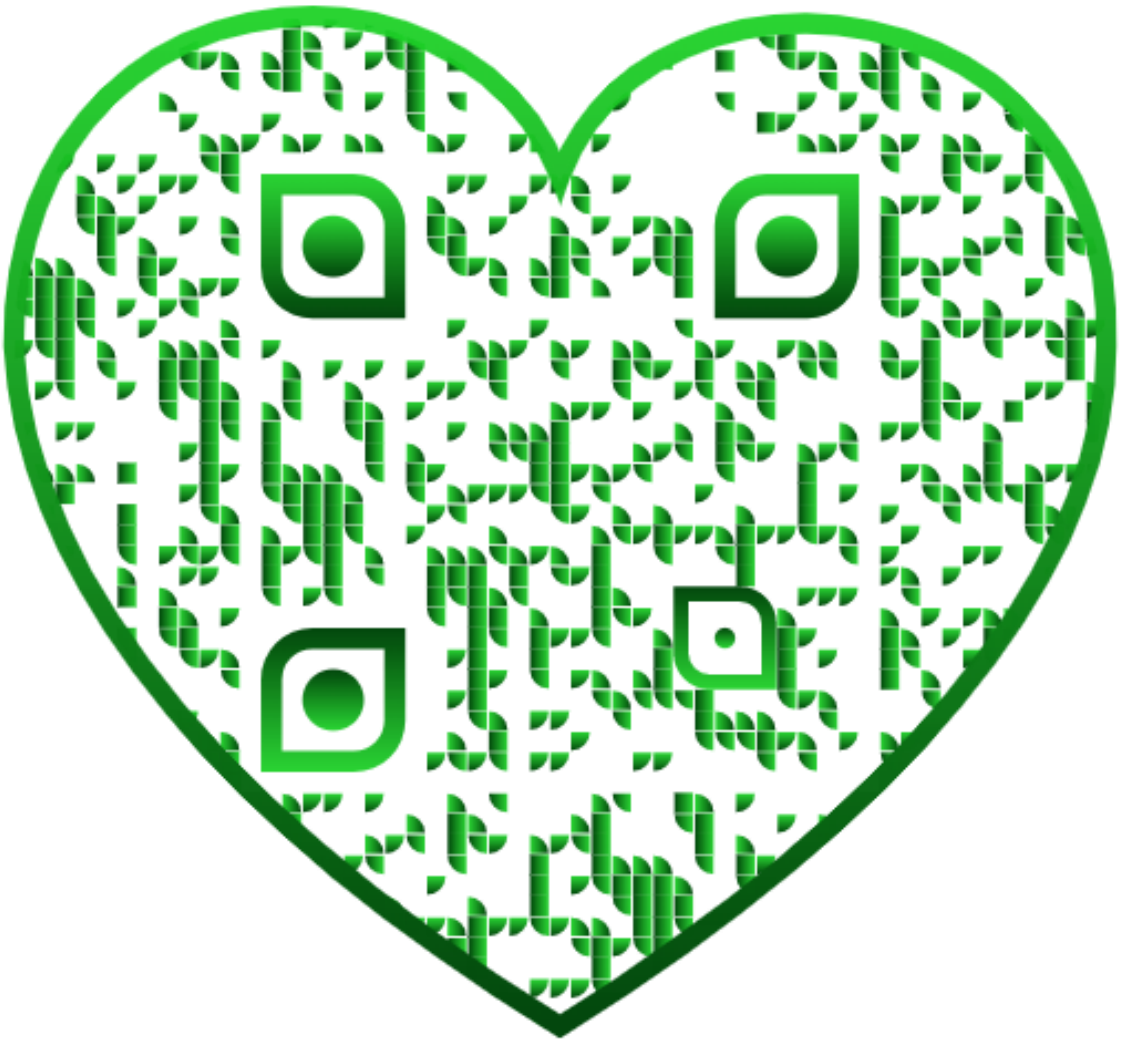
Learn how to manipulate space and get more experimental with your reverbs

SEDIBUS
The Orb spin-off on their latest adventures



MARIA W HORN
Discusses her unique approach to crafting spatial synthesis

WAVETICK
The sample marketplace on its artist-focused ethos



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welcome

Something we're always eager to underline is how the act of going outside your comfort zone, and learning to really *use* all those plugins on your hard drive, can result in new avenues of creative possibility. In terms of effects, reverb typically tends to get short shrift when discussing how to get really experimental. While that other spatial effect, delay, is frequently discussed in terms of its ability to concoct unique flavours, reverb is still often just applied with minimal fuss – slapped on a track to add a touch of room ambience to a recording. Our cover feature this issue cracks open the reverb egg, as our specialist Stuart Adams demonstrates myriad means of making reverbs go further. Guided by science and illustrated with easy-to-understand practical guides, his aim is to take your array of reverb plugins into uncharted territory. Begin your journey into the outer reaches of space from p12.

Keeping in line with our reverby theme this issue, we bring you a conversation with Maria W Horn on p30. The Swedish composer has developed an ingenious workflow that harnesses the impulse responses of real spaces. We also chat with The Orb's spin-off duo Sedibus about the making of their new interplanetary record, *SETI*. Read from p50. We've got another incredible free gift from our friends at Modalics. BitFuzzer melds bitcrusher, distortion and fuzz in one super-modern and straightforward package. Grab your copy from our download page and learn how to use it on p88. Speaking of downloads, we've rustled up two new sample packs for you this month, spanning modular synthesis and the wonky world of glitch. I hope you learn from, and enjoy, this issue.

Andy Price Editor
andy.price@futurenet.com

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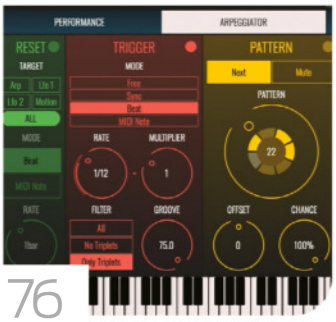

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NEW RELEASES >COMMENT >INDUSTRY HAPPENINGS

Listen better

IK Multimedia's ARC Studio room correction promises a monitor upgrade like no other...

> Last month, IK Multimedia released ARC Studio, a standalone hardware processor for its newly-updated ARC 4 acoustic room correction software that promises to "instantly upgrade" your studio monitors.

To give you a primer on the concept, room correction software compensates for the acoustics of the room your monitors are placed in by analysing the acoustic profile, identifying problematic frequencies and adjusting the output of your monitors accordingly. Everything from a room's size and shape to the furniture in it can dramatically affect its acoustics; room correction accounts for this to achieve a neutral frequency response.

IK Multimedia have been in the room correction business for a while; previous iterations of its ARC software were designed to run on your machine, whereas ARC Studio gives you the opportunity to run the software on a dedicated processor that sits in between your interface and your monitors. There are a number of benefits to this approach, such as eliminating the need to place the ARC 4 plugin on the master bus of a project or introduce additional latency to your monitoring setup.

ARC Studio arrives bundled with a precision MEMS measurement microphone that's used to help measure the acoustic profile of your listening environment through the ARC 4 software and compensate accordingly, delivering "controlled low-end, tighter transients and pinpoint stereo imaging". Separate profiles can be created, stored and instantly recalled for the purposes of different monitors, listening positions and room configurations.

On top of these abilities, ARC 4 now offers expertly modelled monitor emulation, so you're able to reproduce the frequency and phase response of well-known studio monitors from other manufacturers, hi-fi speakers and even mobile devices to get a sense of how your mix will sound on other systems. Something that has often been a laborious final stage process that many don't undertake – but can give a massive indication of how most people will hear your track.

Designed and manufactured in Italy to exacting specifications, the ARC Studio hardware boasts a 120dB(A) analogue input to analogue output dynamic range, 0.5Hz-40kHz frequency response, and ultra-stable clocking. IK says it offers the transparency and accuracy of mastering-grade converters at a far more affordable price.

ARC Studio (including the ARC 4 software and measurement mic) is priced at $/€299.99, but existing ARC users can upgrade for a discounted price. ARC 4 can be purchased on its own for $/€149, or $/€199 with the measurement mic.

ikmultimedia.com

Hoover this one up

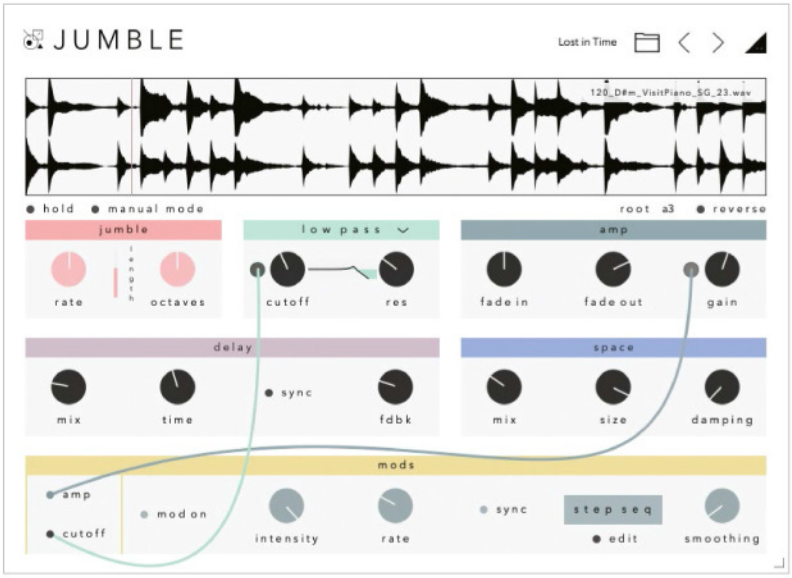
TAL Software recently released TAL-Pha, a new software emulation of the Roland Alpha Juno 2 – a synth primarily associated with the 'hoover sound' that was omnipresent in '90s rave, hard house and trance music.

Bringing together digitally-controlled oscillators with analogue filters and a minimal, knob-free interface. The plugin's architecture mirrors the Alpha closely, with a single PWM-capable oscillator (and additional sub-oscillator) running through two filters, a high-pass and a 24dB/oct low-pass filter that's been given a resonance boost for self-oscillation. It also sports deeper abilities that the original couldn't achieve. TAL-Pha is available now for $80/£63.

tal-software.com

A bit of a jumble

With Jumble, SoundGhost's brand new VST/AU sampler plugin, users can drop in any audio and its sample playhead will shift between random positions within the loop to craft new patterns and ideas in sync with your DAW's tempo. The length of the gate can be adjusted along with the rate that the playhead shifts to a new position. The plugin's amp section allows you to adjust the fade in and out times for each sample snippet, along with a handy gain knob. Additional filters and LFOs make pattern-forming easy, too. Jumble is priced at £79, but available at time of writing for a discounted £49.



soundghost.net

Wavetick: forging an equitable music market

Promising an artist-focused service that gives musicians, beatmakers and producers a free marketplace to trade their sonic wares, Wavetick aims to disrupt the current sample website paradigm in the creator's favour. Let's find out more...

Wavetick
Sharooz Raoofi
Owner and Co-Founder

As a founder of such industry tent poles as Sample Magic, *Attack Magazine*, Audiaire and Principle Pleasure studios, Sharooz Raoofi has already left a sizable footprint on the music production landscape. Whether wearing his entrepreneurial hat or as an artist in his own right (you might be familiar with his hailed alter-ego Principleasure) Raoofi is keen to shake up the mould. This is strikingly true of his latest venture, Wavetick. Alongside a small band of cohorts, Sharooz aims to bring a fairer, more flexible and entirely unrestrictive service for producers, songwriters and beatmakers, creating a platform where they can sell their creations frictionlessly whilst retaining 90% of all royalties. It all sounds fantastic, but we had more than a few questions...

cm: So firstly, can you give us an overview of Wavetick – what is it, and what benefits does it provide to musicians and producers in contrast to the other available marketplaces?
Sharooz Raoofi: "We're really trying to do things differently. One thing I think the world doesn't need right now is another site that's selling sounds and tracks. But there's certainly a lot of unique features that we offer that other people don't. Particularly with the advent of AI, we're really positioning ourselves as an artist-focused and artist-led operation. The first thing to say is that you don't have to pay to be on there. With some folks, you have to pay some form of service fee whether it's monthly or yearly in order to onboard and sell. We don't require card details to be on-site. It's essentially completely free and takes just a few seconds to onboard. The second thing is that we only take a very small platform fee. We take just 12.5%. That number is quite deliberately fixed there as it's lower than Bandcamp which is 15%. We figured it was the lowest possible rate we could do while guaranteeing the service we want to provide.

"From a seller perspective, that probably stands out the most, but once you start diving into the product you'll see that we have quite a unique setup. One thing that's quite unique is that we can offer 'Limited Edition' products. This means that we can offer just one licence to just one buyer. So, in essence you could sell 'ghost produced' tracks on there that others can buy and work with. Or, completely one-off sound-packs that only one person can buy. We can extend this further with 1-of-10, 100 or≈200 editions too, so it's possible to sell completely limited-edition works there. We found this to be quite popular. I spent four and a half years at Splice, and one of the key takeaways was a lot of people are using the same sounds. We are offering that limited edition ability to have something that nobody else has."

You've had quite the career as an industry titan, so what was it that prompted you to start Wavetick now?
"I think it was just examining some of the other products that were in this space, and listening to the frustrations. Not just as someone who is interested in being an entrepreneur, but as an artist myself. Thinking 'what would I gravitate towards?'. And what has the least friction for users. What offers the fairest terms. The big thing that we're doing is taking aim at this idea of 50/50 splits on licence fees. Some of our competitors are still retailing samples, sound packs and production music at 50/50 licence fee splits. That was a major motivator, but also just the overall user flow and experience.

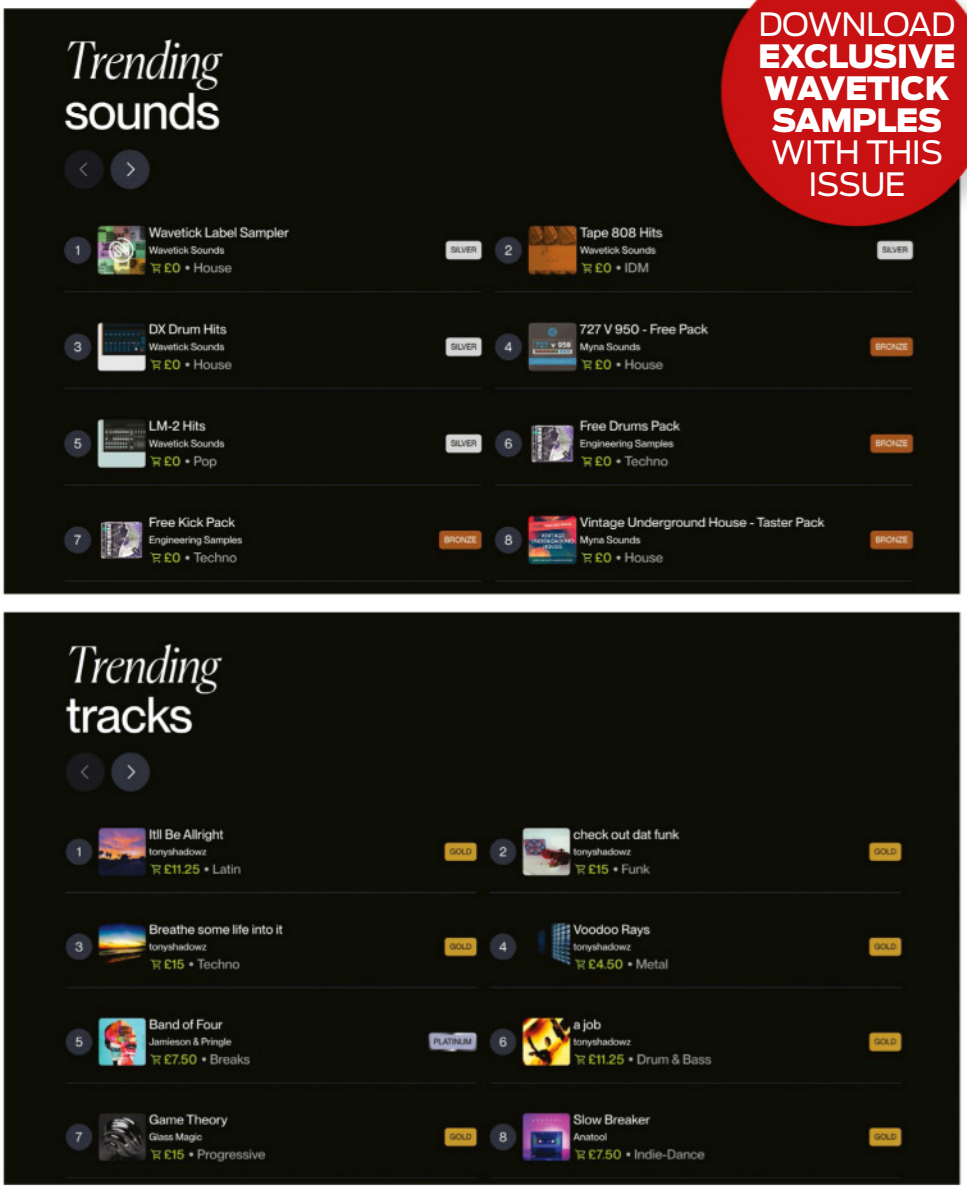
"It doesn't need to be complex, and certainly doesn't need someone to be hoovering up that much of the income. It shouldn't require someone to be paying to be selling. Looking at all these things as an overall health-check of what's going on in our space, was our starting point."

How complex was it to set up Wavetick from a legal point of view?
"The legal part of it was a lot of work. What you're really paying for on our platform – aside from merchandising space – is a legal framework within which you can license one-off beats, tracks and sounds to a user.

> "Platforms don't need to be complex, or for someone to be hoovering up that much of the income"

Legally that's really complex actually. So we've devised these purchase tiers that are coloured. They go from Bronze upwards. The more precious the metal (or colour), the more rights the licensor is being granted. A Bronze-tiered product would essentially be a royalty-free sample pack with the rights granted by beatmaker/seller. A 'Black' beat is something that is essentially owned by us as a platform and is licensed by us to the end-user. As both a seller and a buyer, you can cherry pick exactly what you want and the rights that go with it. The legal work that went with that was incredibly complex, but we managed to break it down in an easy-to-digest way.

"Another issue we had with the beatmaker space, particularly with the rise of NFTs a few years back, was that there wasn't any governance or legality to what was being done. Everybody had to accept that blockchain was this sort of great 'validator' of what was happening, but there really wasn't any legality granting the terms of the licence. We want to be very careful that what we're doing presents security for both sides of the coin. The legal work was a really big part of it, and I'm really glad we got that right."



Wavetick offers a tiered system of rights management, and easily navigable categories

How much insight and analytics do creators get on how their material is being used?
"Well we have a royalty dashboard built in, but in terms of who's using tracks or sound packs, we don't have any after-care so to speak. But for tracks, you enter your CAE/Performing Rights IPI number when you go to sell. When a buyer gets a track from a seller, that CAE information is actually traded. If you buy a track and sample it into your new work, then there's actually a publishing interest for both parties in that work. So the person who has sold it gets 50% and the person that bought it gets 50%. It goes beyond the focus of actually paying a fee to license a track. It gives both parties an opportunity to share in the ecosystem in a much better way. They exchange email information through the CAE.

"It's almost like a peer-to-peer licensing system; we're not really involved in it. We provide our legal terms and our storefront. But if you were to buy a track from me, for instance, and use it in a derivative work (say, rap over it or put a vocal over it) the responsibility of how that new work is registered is the responsibility of the maker and the purchaser."

How big is the team at Wavetick and where are you based?
"There are six of us and we're based between the UK and Los Angeles. We have an in-house sound label – Wavetick Sounds – as well. There's a bunch of stuff on the site that we actually own and we licence that to people who buy it. You can think of that like Sample Magic mk2! We put real effort and care into that, and we can distribute it all over the world. It's a really beautiful thing to see, when we've got sound designers from literally every European territory and across the USA."

How many sample packs are there now?
"45, and we've got a few more in the pipeline too. We're hoping to get to 100 by the end of this year. We're really putting up quite a lot of material. It's become sort of second nature to us because we know so many great sound designers. Going back to one of your earlier questions, one of my main motivations for doing this is that just about

everybody I know is in this space, making great music and sounds, and a lot don't have an outlet to do anything with it. When we first started building the terms and conditions for working with us, a lot of folks just wanted to be paid upfront. And weren't interested in the royalty consideration at all, they just wanted to be beat-sellers. So we just decided to make our own label. That's how Wavetick Sounds was born really."

Do you have any early favourite packs?
"We just launched one today which is actually from the Sofia Philharmonic Orchestra. It's not your usual sample pack for a unique developer like us. It's actually beautiful. It's a selection of live string loops tracked at the Plovdiv Radio Hall. We're selling it for just $25. Another big attribute of our Wavetick Sounds catalogue is that it's significantly cheaper than a lot of other stuff out there. I think it equates to roughly ten cents a sample, which is value for money."

How will you expand Wavetick in the future? Will you offer plugins and software tools also?
"We're making plans to start issuing serial numbers, so with the margin being as low as it is and the platform fee being low, we're hoping to attract some indie software developers. We don't want to end up being a reseller market. The same goes for the artists and tracks side – we're really artist-focussed. But if there is a boutique developer out there that has made a really unique utility that isn't really distributed elsewhere, then that's a perfect fit. It's a frictionless site where anybody can onboard, but we're really interested in those hidden gems. That goes for sounds, tracks and subsequently software. We want to bring attention to those folks who haven't been discovered before."

Are you worried about the threat posed by AI on the royalty-free music space?
"I don't think so. If it was a threat I think it'd be more of a threat to sound packs and stock music. Having spent equal amounts of time both as a sound designer and as a writer of royalty-free music. It's certainly helpful, and there are some brands who'll go to it. But if you're serious about what you do, both in terms of producing and licensing, I think you'll be more inclined toward having some sort of human interaction and an artist focus. I think that's where ours is really beautiful. We're making it very clear that it's made by artists for artists.There is also a major copyright consideration when it comes with AI-generated works. With our storefront it's not really a consideration as there are so many licensing tiers which span from direct licensing from the beat/track-maker to licensing from us as a platform. There's a safety net there – a guarantee that what you're getting from us is legit and isn't something that has shady data-set training, and might incorporate work that is someone else's intellectual property."

> "We want to bring attention to those who haven't been discovered before"

What's coming up next on the agenda for Wavetick?
"We're starting to think and work on the serial number stuff for sure. We are going to be adding way more beat-maker functionality tools in there, so you'll be able to see a lot more detail on usage. I think we want to try and go a little further with what we can offer as a publisher. Hopefully we can add in more opportunities that tie into sync and blanket licensing and things like that. Sync is pretty important right now, everyone seems to want to break into it. If we can help that in any way then we will. We're really keen to expand the creator base to have a more diverse range of people. We want to see more artist-led faces that aren't the norm, we want to offer that level of diversity. Some of our competitors are getting there too, I went to an awesome Girls Make Beats party at NAMM. There are a lot of awesome people out there making incredible sounds and we have to work extra hard as a platform to make sure we're representing those folks on our site."

For more information on the service offered by Wavetick, head over to wavetick.com. *Computer Music* readers can also access an exclusive demo pack at bit.ly/cm333downloads

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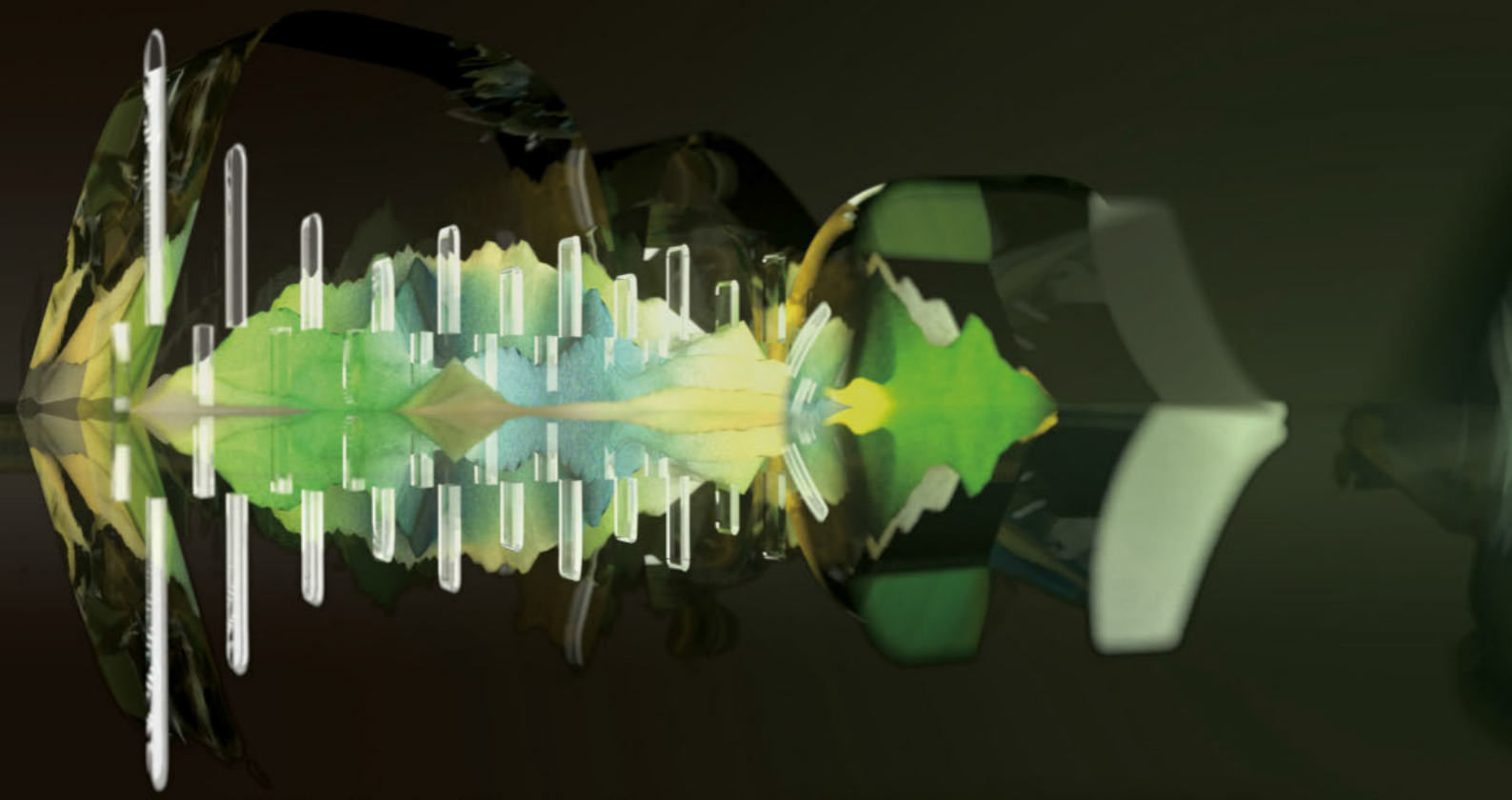

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THE ULTIMATE GUIDE TO CREATIVE REVERB

Reverb has been relied on for its evocative sonic character ever since the first homo sapien warbled expletives into a dark cave. The transformative power of space is still a vital mix ingredient

> As creative music producers we're always looking for ways to catch the listener's interest and attention. We agonise over creating just the right feeling and emotion through melody, harmony and rhythm, and ruminate for hours – days, even – over tiny production decisions that (in all honesty) nobody but yourself will notice.

It is therefore astonishing that, when it comes to reverb, many producers will simply fire up an instance of 'Whatever-Verb', load a preset that sounds nice in isolation, and then pretty-much leave it at that. But, as with most aspects of music production, reverb should always be tailored to the specific situation. A music producer has to consider the style of music, the mood of the song, the instrumentation, the performers... all of it.

Presets can be a useful signpost to help us get into the correct ballpark – for those who are wanting to work that way – but you should never rely on them wholesale; the producer's skill lies in knowing how to adapt to the situation, and knowing how to use the tools available to achieve the desired result. Not to mention the way in which reverb can be manipulated in myriad colourful ways.

Reverb plays on the brain's intrinsic ability to infer information about an environment from the reflections that accompany the sounds we hear. We do this unconsciously and automatically, often unaware that the reverb is there at all unless we actively listen for it.

When used well, reverb gives us a mainline into our audience's subconscious, allowing us to influence their aural experience at a subliminal level. When not fine-tuned to the situation, though, the results can be mushy or jarring, obstructing rather than enhancing the mix.

Essentially, there are two uses for reverb. It can simulate rooms and acoustic environments, bringing realism and depth to your productions. It can also be used as a sonic enhancement or creative effect that blends with and becomes an inseparable part of other voices or instruments within a mix. Allow us guide you through...

Reverb influences our perception of depth and distance, allowing us to create quasi-3D stereo images

Reverb types

Before you get creative, it's important to understand how different types of reverb work

There are four different ways in which reverb is created in the studio. The first to be invented, the echo chamber, isn't a simulation but rather a room in its own right. The concept is simple: place a speaker in a lively acoustic space, play a signal through it, and collect the resulting reverberations with a microphone. Despite creating *actual* reverb, echo chambers are impractical – fun if you get the opportunity, but artificial solutions make life easier...

Lexicon's reputation for reverb excellence started with its original digital algorithmic reverbs, many of which are now immortalised as plugins

Electro-mechanical reverbs – plates and springs – were the first such solutions to come along, but, whilst characterful, their usefulness is limited by the scanty control one has over the reverb sound. The same cannot be said of digital 'algorithmic' reverb, which creates big stacks of delay lines to simulate what happens when sound waves bounce and scatter around a room, and so offers many ways in which to influence the reverb effect.

Convolution reverb imprints onto a signal the acoustic characteristics of a so-called 'impulse' recording. Put another way, the impulse acts like a sample of an environment's acoustics, and a convolution reverb processor can use that sample to recreate those acoustics. This makes convolution reverb infinitely flexible in theory but, like sample-based synths, can also lead to accusations of same-y-ness. The controls on offer tend to be less comprehensive than algorithmic too, and tend to work by manipulating the impulse, so in reality convolution reverb is not as malleable as algorithmic.

Room acoustics

Individual reverb processors – even those of the same type – will offer their own particular selection of parameters for controlling the effect. This can be confusing, but what all processors have in common is that the parameters are designed to in some way relate to what happens in a real acoustic space. To get the best out of any reverb, then, it is important to have an appreciation of the real world acoustics that are being simulated. Let's have a brief recap...

Imagine a sound source within a room. Sound waves propagate from the source, and will at some point encounter a surface that will partially absorb and partially reflect the waves. The initial reflections, referred to as early reflections, will propagate, and the process will repeat, building to a chaotic wash of reverberations until the reflections fully

Convolution reverbs can simulate any space for which you have an impulse recording, but are not as malleable as algorithmic

dissipate. Standing in amongst all of this is the listener, who hears three distinct elements: the direct sound waves, the early reflections, and the reverb wash (or tail). The listener's ears and brain detect the timing, timbre and phase differences between these elements, inferring from them details about the room's characteristics.

How it works

In a reverb processor, the room size or type (algorithmic), or impulse (convolution), determines the basic acoustic characteristics of the effect. For example, larger spaces will tend to generate longer delays between direct sound, early reflections and tail, and that tail is likely to be dense and dark. Conversely, reverberations in a smaller space will tend to be snappier, zingier and less dense; the sound waves being dissipated more quickly due to the rapidity of reflections.

The overall decay time or reverb time is a factor in the apparent size of the room, but is also indicative of how the room's walls reflect and absorb sound waves: longer decay times equate to greater reflectivity, shorter times equate to greater absorption. Also, surfaces tend not to reflect and absorb all frequencies evenly, affecting the brightness or darkness of the reverb tail as it decays. Most processors include EQs, filters and/or damping to help simulate all these surface characteristics.

The balance between the direct sound and the reverberations determines how near or distant the source appears to be, as does the delay between the direct sound and early reflections. This delay is usually referred to as pre-delay, and increasing this will both make the source appear closer, and make the room seem larger. Increasing the delay between early reflections and reverb tail further influences the sense of size, and can also give the impression of additional surfaces existing between the source, walls and listener (a floor, for example).

The warp factor

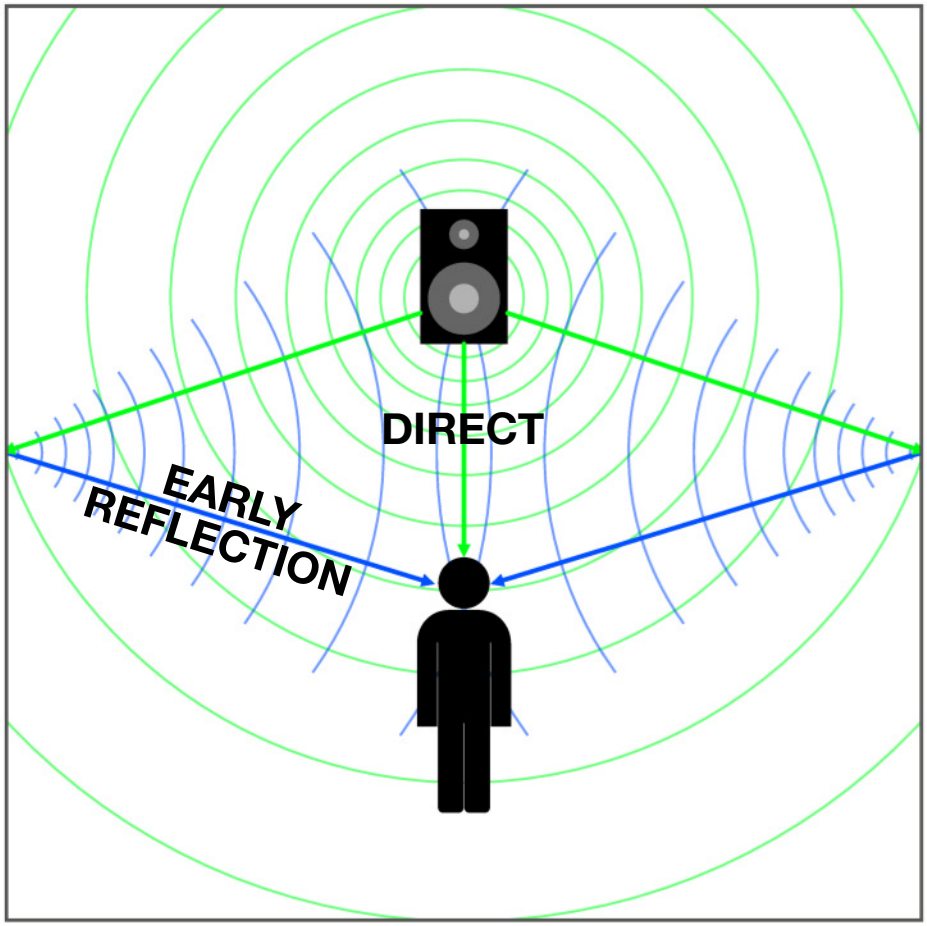
Once we've got the hang of using algorithmic reverb for room simulation we are well set up to start getting more creative with our virtual rooms. After all, the better we understand the relationship between reverb parameters and real world acoustics, the easier it is for us to imagine ways in which to mess with those parameters to create spaces that couldn't exist in reality, and which therefore bring unusual and unique elements to our music.

Convolution reverbs offer a different route to creating strange and other-worldly spaces in which to stage our music. Because the effect is entirely based on the impulse recording being used, working with unconventional and unusual impulses – or even using random noise or pitched audio snippets in place of impulses – can yield all sorts of weird and wonderful results. So, gather as many impulses as you can find and start experimenting!

Creating depth and distance

We will often want to place several, or even all, of the instruments of a mix into the same acoustic space. In this scenario it makes sense to insert a reverb plugin on its own channel, set it to 100% wet, and use channel sends to route instruments to it. Each channel's send level will determine how prominently it contributes to the overall reverb, thereby giving some control over the channel's apparent front-to-back depth – the more 'reverby' an instrument, the more distant it will seem.

We can take this up a level by simulating the loss of top-end clarity and increase in woolliness experienced as sounds move further away. To do this, send the source channels to the reverb via FX/Auxiliary tracks that have EQ processors inserted. Use the EQs to reduce high frequencies and/or increase lower ones within the send signal, based on how distant you wish them to appear.



Longer delays between the direct signal and the early reflections give the impression of less distance between source and listener

> Step by step

1. Simulating and warping a real room

1 > Let's explore the various reverb parameters and how they relate to room acoustics. Start by loading an existing project into your DAW. If you don't have anything suitable then you could grab one of your DAW's demo projects, or download something from a multitrack library such as Cambridge MT [cambridge-mt.com/ms/mtk]. Disable any existing reverbs and reverb sends used by the project.

2 > Create an FX/Auxiliary track, name it 'Room Reverb', and insert an algorithmic reverb on its channel. Set the wet/dry balance to 100% wet. We're using Steinberg RoomWorks, included in Cubase and Nuendo, because it exposes lots of useful parameters and names these clearly. You can use your own choice of plugin, but aim for one with a similar selection of parameters.

3 > Work through the parts of your project, creating sends to the Room Reverb from any channel that you wish to place into the simulated space. Be sure to use post-fade sends so that the balance between channel level and send level always remains consistent. For now, keep your Room Reverb channel quite loud so that the effect is obvious.

4 > Our song will suit an intimate live venue sound, like a smallish club (follow our lead on this for now). Set a middling reverb/decay time of 1.2s then adjust the room size/reverb type, noting how this affects the reverb's snappiness, brightness and density. Starting and stopping playback helps these details stand out. We've settled on a value of around 33% of the maximum room size.

5 > Pre-delay affects the perception of room size, and of distance between sound source and listener. It also has the happy side effect of creating separation between the direct signal and the reverb it creates, improving clarity and left-right positional accuracy lost to the smeariness of reverb. Adjust the pre-delay, aiming for space and clarity whilst avoiding any echoey slap-back effects.

6 > Applying EQ to the reflections being generated by the plugin helps us simulate different wall materials – this is typically called 'damping', but may also be presented as conventional EQ or filters. Our imaginary club's textured walls are more absorbent at higher frequencies, so we're damping high frequencies above 3.5kHz. It also has lots of bass-boosting corners, so we're boosting low frequencies below 230Hz.

7 > The low damping boost has made things a bit muddy – fix this with an input filter/EQ cut below 350Hz-ish (insert an EQ plugin before the reverb plugin if needed). Adjust the reverb/decay time to get things just right. Work through the channel send levels, adjusting how much each contributes to the reverb (more reverb equals more apparent distance).

8 > Go to the Room Reverb channel's fader and adjust its level, balancing and blending it with the dry mix. A good way to get this balance right is to regularly mute and unmute the reverb channel for comparison. When the reverb is kind of imperceptible until it is muted, at which point you clearly notice its absence, then you have the balance about right.

9 > Experiment with your plugin's parameters to help you learn what each actually does, and go ahead and add other processors to the Room Reverb's insert chain, placing them both pre- and post-reverb to compare their sound and impact. We've created extra richness with some post-reverb tube overdrive to add extra harmonics and complexity.

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Reverbs that are an intrinsic part of an instrument's sound, such as a guitar amp's built in 'verb, can be additionally included in a wider room simulation

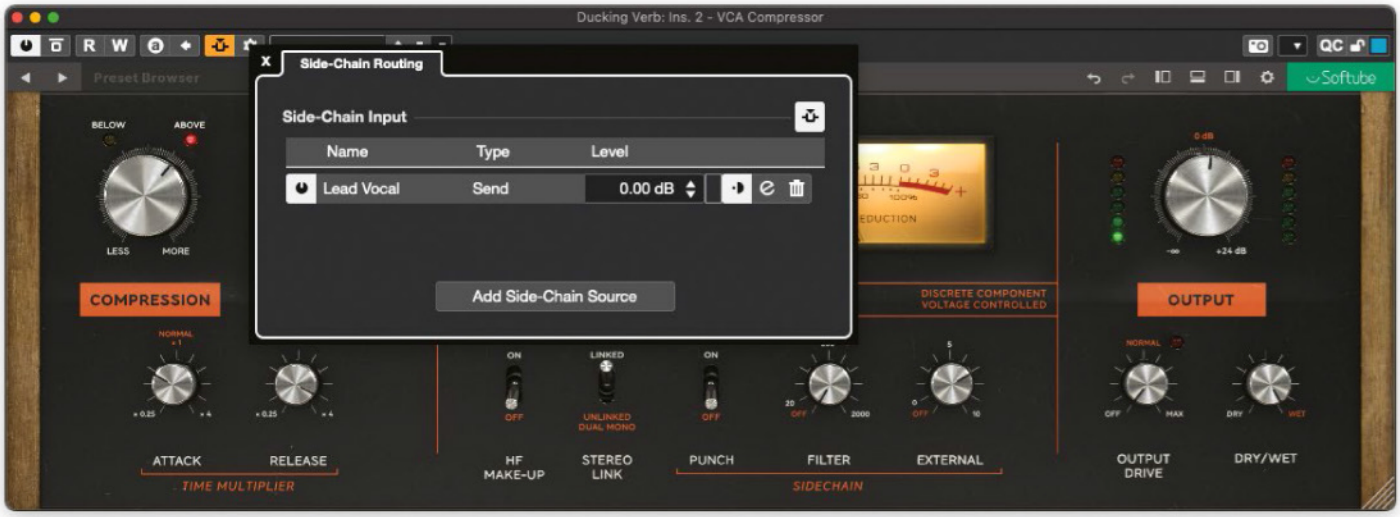
Deeper reverb manipulation

Now you know the basics, let's delve into the more complex attributes of reverb control for our tracks

Placing the instruments and voices of a mix into a consistent space does not preclude us from adding additional reverbs to specific parts and tracks. This is particularly true when a reverb is being used as enhancement rather than for simulation purposes.

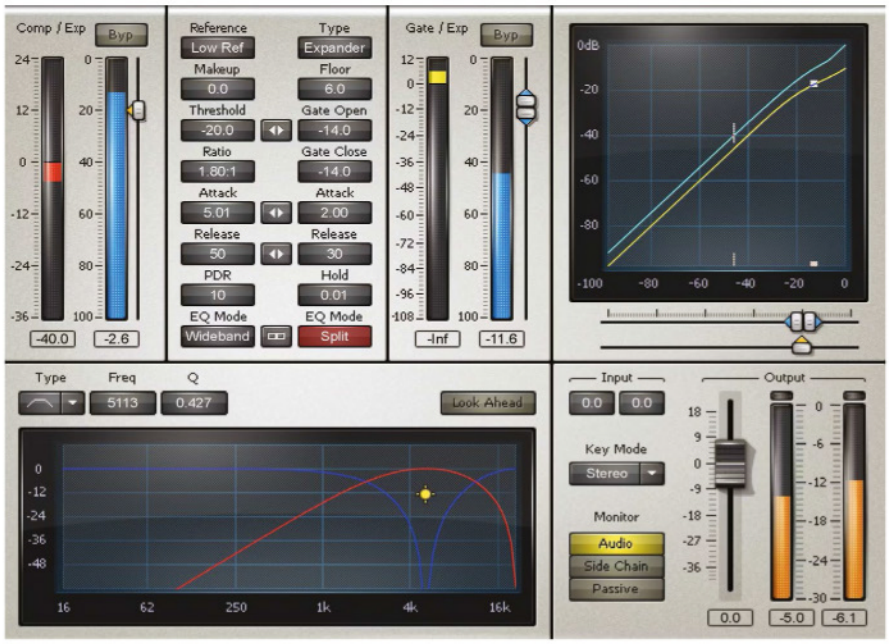
For example, you may wish to add some spring reverb to a guitar track to help emulate a classic guitar amp sound. That reverb becomes an intrinsic component of the guitar performance, and would be present if performed in a live venue, and so it absolutely makes sense to also include it in the overall room reverb.

It is also not uncommon to apply more than one reverb to a single track when creating track-specific enhancements and/or special effects. This is especially true of vocals and other lead



Ducking is a classic compressor-based effect that can be very useful when applied to reverb

Spring reverb has long been a favourite of guitar players



Waves C1 combines gating, compression, dynamic filtering and much more in one plugin

lines; a good example being having both an up-front gated reverb and a more subtle, spacious reverb on the same part. In this case, however, we may not want to blend the 'enhancement' reverb with a shared room 'verb.

Stacking and combining reverbs in these ways is just fine, but always think carefully about the signal routing, and work out how to create such routings within your DAW (some make it easier than others). If you're happy to have the reverbs operating in series, then inserting them on the same track is normally sufficient, but more often you will want to run the reverbs in parallel. This can become a bit elaborate, requiring separate FX/Auxiliary tracks to host each reverb plugin, and various sends and submixes to route and combine the signals.

Sidechains and ducking

There are various ways to use dynamics processors in conjunction with reverbs. The classic gated reverb effect is probably the most familiar, and is something many reverb plugins can produce internally. This often limits us to a specific reverb algorithm, which can be restrictive in terms of the tone and timbre of reverb. It's therefore useful to know how to create gated reverb using *any* reverb processor – we'll show you how in a moment.

> You may wish to add some spring reverb to a guitar track to help emulate a classic guitar amp sound

Another common technique involves combining reverb and a compressor-based ducking effect. The ducker controls the volume of the reverb, reducing it when an instrument/voice is sounding and bringing back in the spaces between phrases. This works particularly well with longer 'verbs, keeping them out of the way when the part is doing something important, whilst still giving us luscious reverb tails afterwards.

Both techniques place the dynamics processor after the reverb, but we want these to respond to the dry signal, not the soft and smeared reverb. This means configuring the dynamics processor to key from an external sidechain fed from the dry, reverb-free source track.

Reverb meet delay

When it comes down to it, reverb is, fundamentally, a delay-based effect, but that doesn't mean that we can't combine it with an *actual* delay effect. Again, it is important to consider the routing when doing this. If we want each delay repeat to produce its own bloom of 'verb then we can insert both delay and reverb (in that order) on the track being processed. If, however, we want delay and reverb to be

separate then each needs to be applied via its own independent FX/Auxiliary track.

It's less common to place a delay in series *after* a reverb, because all it's really then doing is extending the reverb time. One exception, however, can be delay-based modulation effects such as chorus and flanging, which we may want to apply to the reverb itself. That said, placing such effects pre-reverb tends to give a richer reverb without the modulation effect itself becoming noticeable.

The joy of automation

Automation and modulation of parameters can bring any effect to life, but with reverb it gives us some particularly rewarding possibilities: darker 'verb in quiet sections and brighter in louder sections; rhythmic reverbs; or near-to-far depth movement effects, for example.

Don't forget that in situations where a reverb is chained with other processors, we can also influence the characteristics of the reverb by automating or modulating the parameters of those other processors. In complex chains, and where multiple parts are feeding the same reverb, those processors may be on tracks other than the one hosting the reverb.

> It's less common to place a delay after a reverb, because all it is really doing is extending the reverb time

Most reverb plugins will expose all of their parameters for automation, but we do have to be cautious about which we modify in real time because some will not change smoothly, and may even cause the reverb to mute or stutter while the new value is loaded. Also, keep in mind that adjusting delay-time-based parameters can cause a pitch-bending effect while the value is in motion, much as it can when it comes to a standard delay processor.

As always, experimentation is key. Try out new things, think of imaginative ways in which a reverb could be shifted and warped, and then explore different methods for realising that imagined effect.

Placing modulation effects either before or after a reverb can thicken and enrich the reverb sound

Most DAWs offer a means to convert audio events into MIDI notes – the secret to clean drum 'verb

Creating spill-free drum reverb

When mixing, we ideally want every part to be fully isolated so that any effects we add impact only the parts we want them to. This isn't so easy when it comes to a drum kit, with every individual drum mic picking up spill from the rest of the kit. As a result, if we want a specific reverb effect on a specific drum, some of that reverb will also find its way onto the rest of the kit. A clever solution is to use samples to drive the reverb.

Convert the individual drum track to MIDI notes – most DAWs can do this one way or another – then copy a clean, single drum hit from the source recording (or find a similar-sounding sample) and load it into a sampler. Tweak the MIDI part for accuracy if needed, and use it to trigger the sampler. Run this through a 100% wet reverb for clean, spill-free drum 'verb.

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To create a genuine reverse reverb we have to mimic in our DAW the way it was originally done by those music production pioneers

Reverb effects

While we can manipulate those regular sounding 'verbs with ease, it's time to explore how to make all manner of reverb-based effects...

So far we've been looking at using reverb as reverb – that is, using it to create acoustic spaces and interesting ambiences – and we've also seen that there are all sorts of creative things we can do within this context. But there are also various ways in which we can harness reverb as a sound source or instrument in-and-of itself, and use it to create special effects that have a distinctive character all of their own.

A good example is a strange resonance trick that many (but not all) algorithmic reverbs can pull off. If we combine a small reverb/room size with longer decay times, and get the wet/dry balance set just right, what we can end up with is a resonant, reverberant, somewhat pitched, somewhat comb-filtered sound that would be tricky to create any other way.

Adjusting other reverb parameters, such density, spread and damping can impact this effect in useful ways, and adding modulation creates phaser-style effects. What we're actually doing here is a bit of Karplus-Strong string synthesis, with the reverb acting as a resonator and wave-guide – sound designers take note!

Not all algorithmic reverbs can do this, and those that can will tend to create different results, so try it with a number of different plugins to see how each responds.

Convolution reverbs can also create fascinating filtering and colouration effects (convolution is, after all, used for such things as well as for reverb), so remember to hunt out and experiment with unconventional impulse recordings to see what results they produce.

The classic Yamaha VL1 used Karplus-Strong-based modelling in order to emulate real instruments

BLEASS Reverb's freeze function and X-Y pad let you play the reverb almost as though it were a synth

Baby Audio's sublime Spaced Out melds reverb and delay and allows you to create all kinds of awesome spatial effects easily

All rise

Originally, in the pre-digital days of tape, reverse reverb was created by swapping the tape reels so that everything played backwards, recording the ensuing reverb to the reversed tape, then putting the reels the right way around again. On playback, the reverse reverb fades in and builds towards the sounds that actually spawned it, just as it should do.

Many reverb plugins offer reverse reverb effects and these can be useful, but they're not the real deal because the seed sound always occurs before the reverb itself (plugin developers having not yet cracked time travel!). Creating a *real* reverse reverb in a DAW is easier than it was with tape, but it's still a bit of a faff.

Reverse reverbs are, of course, just one form of riser – those characteristic sonic build-ups that can add so much drama and anticipation to a song. Different seed sounds and parameter settings result in different reverb sounds, making your average reverb plugin a great tool for creating such risers.

Freezing and morphing

Some reverb processors offer a fun feature that freezes and sustains the currently-sounding reverb tail. Automating this is a great way to create rhythmic reverb effects, and mapping it to a button on a hardware controller, along with mappings to other reverb parameters, gives us a way to play the reverb in a creatively spontaneous manner, almost as though it were a synth.

There are a few reverb processors that deliberately lean towards this sort of usage, such as Baby Audio's Spaced Out and BLEASS Reverb. Both have freeze buttons, and both include X-Y pad-style controls that warp and bend the reverb settings in different ways, perfect for creating risers and special effects.

Reverb pads

Sound design thrives on access to lots of timbrally complex sounds to wrangle through a synthesis engine, and reverb processors are a good tool for creating such sounds.

Steinberg Padshop Pro: an excellent granular synth plugin

Here, we're taking advantage of the fact that reverb tails are timbrally rich and complex, but we can somewhat control that complexity via choice of reverb processor, parameter settings and by carefully choosing our seed sounds.

We could, for example, create a reverb tail that is pitched to a particular note or set of intervals simply by triggering it with an instrument playing that note or chord. Similarly, using noise bursts or drum hits lets us create atonal sounds to synthesise with.

Any sound can provide a rich seam of tones for a granular synth to work with. Reverb tails are no exception

Granular synths are best at taking advantage of this sort of sound source. As we know, any sound or recording can provide a rich seam of tones for a granular synth to work with, and reverb tails are no exception. What makes them work particularly well, though, is that we can pre-seed these tails with pitches and timbres that we like, and even bring in variations whilst the reverb is sounding. What better for breathy and floaty synth pads?

Alternatively, if the reverb is seeded with a drum or other atonal sound, a granular synth can turn this into a unique noise pattern that can be layered with other sounds, used as the basis of new and original drum and percussive sounds, or even used as a convolution impulse.

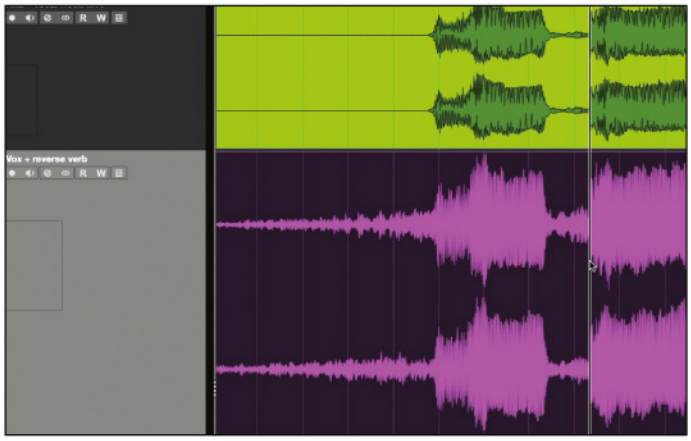
> Step by step

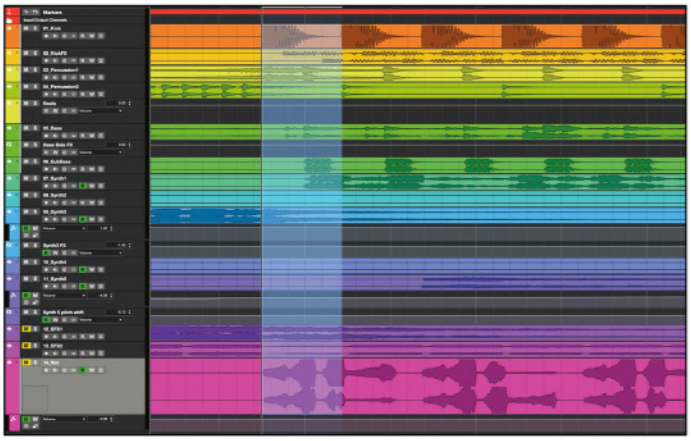
2. Creating filters, reverses, risers and pads

1 > Let's start by creating some interesting filter effects. Import a drum loop or other sound source into a new DAW project, insert an algorithmic reverb, and set its wet/dry balance to 50%. Choice of reverb has a big impact on both the method and results here, so you may need to experiment to get the effect to work.

2 > What we are essentially trying to do is create a controlled feedback situation. A small room/reverb size combined with longer decay times tends to do it, and some processors also have a feedback setting which does the trick nicely! Pre-delay can have an influence here, too. In short, experiment with parameters and plugins, and you'll find a whole bunch of interesting filter effects.

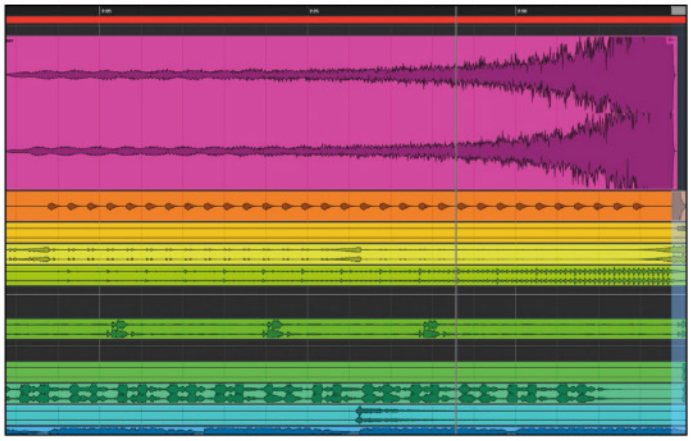
3 > Let's make a real reverse reverb now. Create a new project and import a vocal or other lead part. Insert a reverb on the vocal and set up a pleasing, medium-to-long effect. When happy, use your DAW's audio processing tools to reverse the reverb part – not all DAWs can do this, so use a free audio editor such as Audacity if needs be.





4 > Import the bounced audio to a new track if your DAW didn't do this automatically, then reverse it once again (or, if using Audacity for processing, reverse and then import to your DAW). Carefully position the reversed bounced part so that it lines up properly with the unreversed original part, then mute that dry vocal's track. You now have true reverse reverb!

5 > Risers use similar techniques to reverse reverb, but tend to be less of a faff. Open a project and find a point in the song that would benefit from being preceded by a riser. Export a short segment of audio from this point – sufficient to seed the reverb, and including any tracks that you want to include in the riser sound.



6 > Create a new project and import the snippet you just exported. Insert a reverb, set it to 100% wet, and create a sound with a long tail and plenty of character. Add other plugins for more sonic interest if needed. When done, export the results, import this new file back to the original project, reverse it and position it.

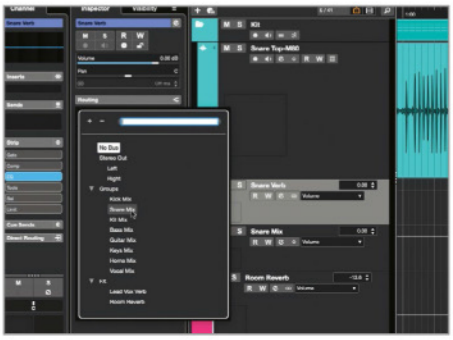
7 > Let's now create a synth pad. Open a new project and create a new instrument track using an synth or sampler that supports granular playback. We're going to use Arturia Pigments, but you could use NI Kontakt, Steinberg Padshop, BLEASS SampleWiz 2, or any one of numerous premium and free granular synths should you not already have something suitable.

8 > Import into your instrument into the unreversed riser sound you created in step 5. Use the synth's sample and granular editing controls to hone in on a pleasing portion of the source sound. Build up a patch for the sound by adding a pad-like amp envelope, and modulating the granular position time to create timbre shifts.

9 > Finally, let's play with some freezing. Import a drum loop into a new project and insert a reverb that has a freeze function – we're going to use BLEASS Reverb. Set up a nice 'verb, add an automation track for the freeze button, and create a rhythmic automation pattern. You could then enhance the effect by automating other parameters.

> Step by step 3. Combining reverbs with other effects

1 > We'll start by combining reverbs on a snare drum. Load a project containing multi-tracked drums and solo the snare, or create a few bars of a snare drum pattern using a sampler or drum instrument. Add EQ and dynamics if needed. Also create an FX/Auxiliary track, label it 'Room Reverb', and configure a room simulation reverb on it.



2 > The actual snare of a snare drum – that is, the springs on the underside of the drum – are essentially a reverberator, hence why reverb works well to reinforce snare sounds. Create a new FX/Auxiliary track named 'Snare Verb', and a new Group/Submix track called 'Snare Mix'. Route the main snare track and Snare Verb outputs to Snare Mix.

3 > Insert a reverb on Snare Verb and create a send to it from the main snare track. Create a snappy 'verb with a bright tone, medium decay time and 100% wet mix (you can tweak the reverb sound later if you wish). Insert a gate after the reverb, and configure it to use the source snare track as its external sidechain source.

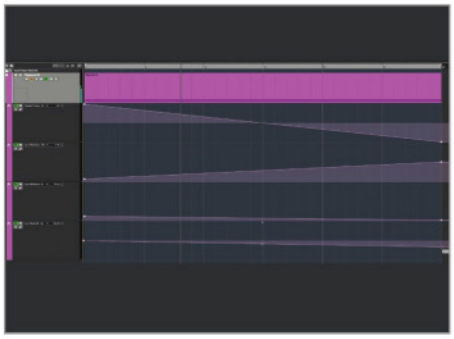
4 > Set the gate's threshold such that it lets through a burst of reverb each time the snare is hit, and use the gate's envelope settings to shorten and shape the reverb tail. Adjust the reverb settings too if needed. Balance the Snare Verb level with the source track, and create a send from Snare Mix to Room Reverb. You now have stacked reverbs configured and routed.

5 > Switch to a project that includes a vocal (or lead) line. As before, create a Room Reverb track, an FX/Auxiliary track for processing the vocal (Lead Vox Verb), and a Group/Submix track that combines the vocal track with the vocal effects track (Vox Mix). Create a send from the source vocal track to the Lead Vox Verb track.

6 > Insert a long-ish, 100% wet reverb on the Lead Vox Verb track. Insert a compressor after the reverb and configure it to key its sidechain from the source vocal track. Configure the compressor threshold so that the vocal exceeds it, and adjust the ratio and envelope controls for the desired strength and smoothness of ducking. Rebalance the effect track with the dry one.

7 > In this instance, we want the main vocal track to be included in the overall room reverb but not the ducked reverb effect, so create a post-fade send from the source vocal track to the Room Reverb track. Work on the balance between the dry track and the two reverbs, adjusting the relative levels until everything sits together nicely.

8 > Now for some quasi-3D positioning. In a new project, load a nice synth patch and disable any effects it uses. Insert a high shelf EQ at around 2.5kHz, followed by a reverb. Create a pattern or arpeggio, and open automation lanes for (or create modulators to control) panning, reverb wet/dry balance and pre-delay, and high EQ cut/boost.



9 > Create a left-to-right curve on the panning lane, a 15% to 60% curve on the wet/dry lane, a 50ms to 0ms curve on the pre-delay lane, and a 0dB to -8dB curve for the EQ high frequency lane. Hit play, close your eyes, and listen to how the sound source moves from close to distant as it pans across in front of you.

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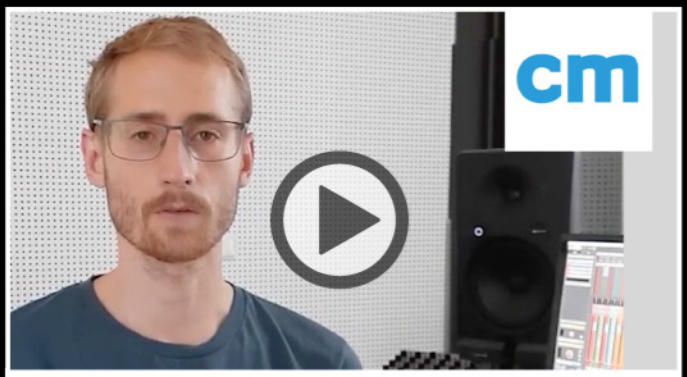

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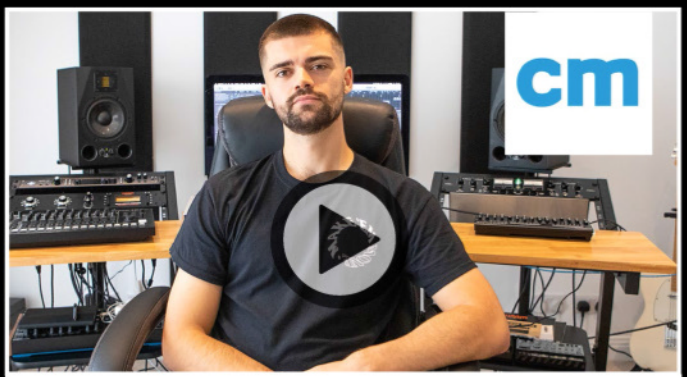

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14 creative reverb pro tips

Take our advice, and become a reverb master in no time

01 ALWAYS EQ

EQing the tracks and parts of a mix is second nature, but far too often people will assume that because a dry part is EQ'd, there is no need to EQ any reverb we've attached to it. Wrong! Adding EQ to reverb is vital, letting us remove low-end muddiness, bring out the top end sparkle and sheen, and create spectral space for the dry signal to poke through, thus aiding with overall clarity.

Adding EQ aids with the overall clarity of your piece

02 USE PRE- AND POST-FADE SENDS

The channel sends we use to feed signals to a reverb plugin inserted on another track can be either pre-fade or post-fade. Typically we use the latter, because this maintains the send level's relationship to the main channel level. Sometimes, however, it is preferable for the send level to remain constant no matter what, such as when using it to trigger a ducking compressor. Using pre-fade sends also makes it easier to manipulate the balance between send and channel levels when creating front-to-back depth effects.

03 FLIPPING STEREO

We may often want a particularly spacious reverb on a part, but this can then interfere with the part itself, making it less distinct and prominent. A simple solution is to use panning to separate the dry and wet signals, shifting one leftwards and the other rightwards. This isn't a realistic or natural effect, but it can sound good, especially on guitars, keys and synth-comp parts.

Use panning to seperate the dry and wet signals

04 VINTAGE 'VERB

Some plugins are designed to give a vintage sound. As for the rest – well, we can make them sound vintage ourselves. Configure the reverb as a send effect, and add some tube- or tape-style saturation and an EQ before the reverb plugin itself. Use these to create a gritty sound with rolled-off bass and treble. You could also add a bit-crush distortion to bring the signal driving the EQ down to 8- or 12-bit quality.

Worried your plugin isn't old-skool enough? Never fear, any plugin can be made to sound vintage

05 MID-SIDE REVERB #1

Applying and mixing reverb in the mid-side domain offers some fascinating possibilities. If working with a stereo source, applying a shorter reverb to the mid channel and a longer one to the side channel helps the central focus of the sound to remain clear and distinct, but the overall stereo width and spaciousness can be enhanced.

06 MID-SIDE REVERB #2

Another option for mid-side reverb, especially useful when working from a mono source, is to route the source track to two FX/Auxiliary tracks with reverb inserted on each. Then, using a plugin such as Voxengo MSED, mute the mid channel from one reverb and the side channel from the other. You can then create different widths and focusses in mid and side channels.

Mid-side 'verb is useful when working from a mono source

07 GETTING YOUR DUCKS IN A ROW

We have shown you how to create a ducking reverb, but how about having two running in parallel? When configured just-right, this lets us inject short, snappy ambience between words and transients, and bring in longer reverb tails at the end of passages and phrases.

08 COMPRESSING AND EXPANDING

Using a ducking compressor reduces reverb level whilst a source part is sounding, so it follows that an expander does the opposite, making sure the reverb is audible during phrases and is taken away afterwards. Taking it a step further, combining ducking and a 'rising' reverbs as a parallel pair lets us morph automatically between two different reverbs based on performance dynamics.

An expander can make sure your reverb audible

09 FX CHAIN ROUTING

Often, the first step in creating an interesting or creative reverb setup is to think through how we're going to route the signals to achieve it. This often involves adding sends, multiple FX/Auxiliary tracks for processing those sends, and Group/Submix tracks for recombining signals. Always think this through logically, keeping in mind which signals need processing where, what the correct wet/dry balance of your various plugins needs to be, and how the various elements will be mixed or balanced.

10 PER-TRACK PRE-DELAY

When feeding multiple instruments to the same room simulation reverb we can only make limited use of pre-delay's depth effects because the pre-delay is being applied within the reverb. We can fix this by setting the reverb's pre-delay to zero, and then routing each source instrument via its own FX/Auxiliary channel with a short, 100% wet delay inserted on it. That is, the delay processors act as independent pre-delays for each instrument.

11 GATING FROM OTHER TRACKS

As we've seen, the similarity between reverb and a snare drum's snare allows us to enhance our snare drum sounds in all sorts of different ways. One trick we particularly like is to add a bright, prominent and washy reverb to a snare, and to then gate this reverb using a feed from the kick drum. This gives a sound that is best described as the snare being sucked into the following kick.

Gating a snare drum hit to a kick drum produces a sound much like the former is being sucked into the latter

Why not take your sound in interesting new directions by adding different reverbs to different frequenct bands?

12 MULTIBAND REVERB

Applying different reverbs to different frequency bands can open up all sorts of intriguing possibilities. To do this, create an FX/Auxiliary track for each frequency band you wish to create, and insert low- and/or high-pass filters onto each track to define the frequency bands. Adding a reverb plugin after each filter then lets us apply a different-sounding 'verb to each band.

13 VARIED ENVELOPES

Some reverb plugins feature global envelopes that apply a final layer of amplitude shaping to the sound, and these are particularly good for use in a multiband reverb setup. Dial in the same reverb settings for each band's 'verb, but give each band different envelope settings. For example, we could configure the low band to have the slowest attack, the mid band a faster attack, and the high band to have the fastest attack.

Global envelopes apply a final layer of amplitude shaping to the sound

14 KNOW WHEN TO STAY DRY

You've no doubt heard the musical idiom which goes something like: "The notes you don't play are as important as notes you do play". The same can be said of reverb. Some sources love wallowing in a bath of the stuff, but others work better when kept dry. Also keep in mind that a dry part can be hugely up-front and attention-grabbing when it sits in amongst instruments that have been draped in reverb. **cm**

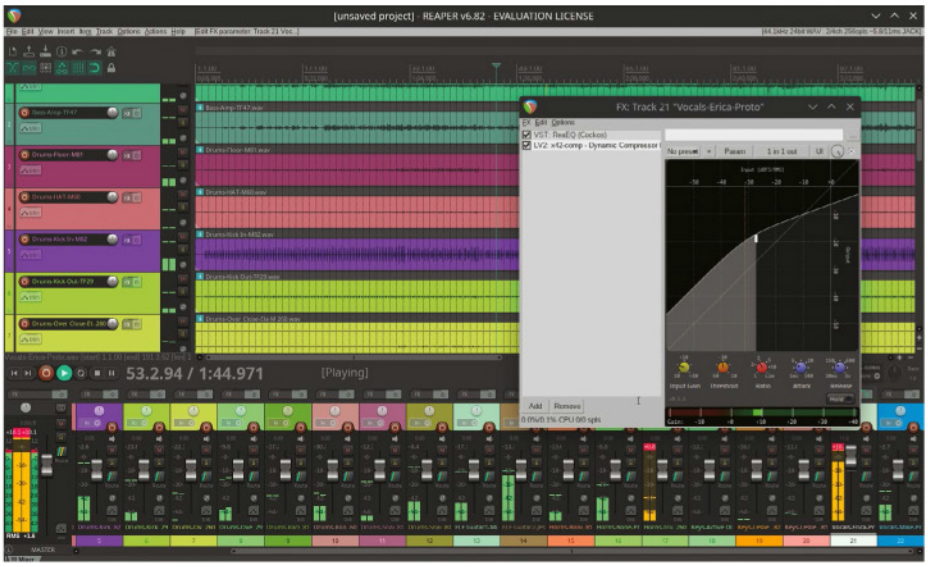
MARIA W HORN

An avid explorer of sound's spectral properties, Maria W Horn's work realises the potential of long-forgotten spaces. Via instrumental textures, room responses, the human voice and a specialised computer language, she synthesises entirely new sonics. We heard more about her processes and the at-times chilling story behind her affecting new album, *Panoptikon*

> Originally operational between 1856 and 1979, Sweden's Vita Duvan prison, in Luleå, in the far north of Sweden, was the country's only 'panopticon prison': an environment designed to facilitate maximum social control through psychological techniques, including isolation and limited human contact. It instilled an omnipresent feeling of being watched via its circular structure. For Swedish composer, artist and electronic innovator Maria W Horn, the prison – now abandoned – was an intriguing prospect.

Maria explains: "As I visited the city in order to research the area for The Arts Biennial, I passed by this circular building that I found fascinating. It turned out to be a former prison. The only 'panoptic' prison in Sweden. As I started working on an installation, the history of the prison gradually unfolded."

Sonic projects of this type are nothing new for Maria: her 2021 work *Dies Irae*, for example, focuses on the acoustics and spectral properties of an abandoned machine hall. The Stockholm-based composer is a well known figurehead of the city's experimental electronic scene, which was underlined by her co-founding of label XKatedral in 2015. Operating at the intersection between audio/visual art and working as a music artist, Maria's latest project was spurred by a fascination with the lives of the people who were held at Vita Duvan. "In this work, I was trying to imagine the experience of the individual prisoner, whose first three years were often spent in total isolation, during which all contact with other prisoners was strictly



Horn is drawn to open-source (or open source adjacent) tools like Reaper for their customisable qualities

forbidden. The silence and uncertainty, the scarce space whose only indication of the passage of time, is the cycles of daylight. How is consciousness affected by the kind of sensory deprivation that an isolation cell entails?

"One passage from the prison books that stuck with me was describing the church service held on Sundays, where the priest was located in the middle of the panopticon and prisoners were allowed to have their doors hatched slightly open, to hear but not see through the cell doors. In this way they could sing along with the hymns from inside their cells, and this is described as a very intense and vivid experience for the inmates, after months of isolation, to get to hear the voices of the other inmates and to share this communal singing experience. This was only allowed for a brief period in the prison's history, as the management believed that the risk was too high that the prisoners could transmit secret messages to each other through the songs.

"I wanted to focus the piece around the idea of vocal transmission echoing through a structure built for psychological surveillance. The individual voices of the inmates are represented by their voices distributed by loudspeakers placed in each prison cell. What if I could use this emotive image and expand it into a vocal cycle? An imaginary choir comprised of ghosts of the Vita Duvan prison."

Hearing voices

Maria refers to her music as spectral, which in part refers to the ideas derived from the 1970s French minimalist music tradition where the composer's artistic choices are based on the harmonic spectrum of the instruments. With the use of electronic technology, this approach can be used to capture the acoustic properties of a room, a technique that Maria used in the prison. "For the last couple of years I've been focused on site-specific composition. An interesting way to adapt the music to a building is to measure the acoustics of the room. I've been making IRs [impulse responses] of different spaces and then used these imprints for spectral analysis. This way I can find out about the resonant frequencies of a building, and use the specific resonance or grits of the acoustics to tune the harmonic universe of the composition.

"I'm using a program called SuperCollider which is a musical programming environment that allows you to program your own synthesis. The good thing is that you can achieve precise control of tuning, texture and timbre when you operate on this level of signal processing. As a starting point for a composition, it is a powerful tool, and the level of customisation allows for very personal synthetic sounds."

Maria continues: "Essentially *Panoptikon* is part of an ongoing musical research of site-specific composition techniques, but there's also an emotional aspect that deals with memory and the idea that events and emotions create inherent memories in a building, object or instrument, and that these 'traces' or 'ghosts' are available if we're willing to listen."

> "Hearing those voices constantly made me hallucinate whispers that weren't part of the composition. It was intense"

With the prison's spectral properties captured, Maria began thinking about the musical components within the concept of *Panoptikon*. "I wanted the piece to have an underlying grid of some sort, so I came up with a 14-chord sequence that runs throughout the whole piece. For the installation, it was in sync with lights, fading in and out, as a way to reflect the passage of time and cycles of daylight in the imagined experience of the inmates. There's a static element to these recurring chord blocks, monotonous just like the days of a prisoner."

On top of this chordal grid, there's the recorded vocals of the four singers [Sara Parkman, Sara Fors, David Åhlén and Vilhelm Bromander] adding melodies and ornamentations, lamenting the passage of time. The resulting tracks; *Omnia Cittra Mortem* (which translates as 'Everything Until Death') *Hæc Est Regula Recti* and *Längtans Vita Duva* are truly affecting pieces. These tracks capture a sense of timeless desperation, fading hope and a longing for human communication. A theme which had further unexpected resonances in 2020 when the installation was first active.

The singers themselves never actually set foot in the prison itself. "I recorded the singers separately, as I wanted to be able to spread and isolate the different voices to the various cells of the prison. The actual setup of the installation within the prison was quite demanding.

"Me and my partner Mats Erlandsson built the installation in winter of 2020 during the Covid pandemic. In December in the north of Sweden there's around an hour of sunlight a

day. Due to Covid restrictions, the Biennale didn't want us to meet anyone. So in a way, we were isolated in the prison for two weeks. I went a bit crazy during that time." The intensity of both Covid and developing the prison installation in tandem, put Maria in a state of mind not too dissimilar to the prisoners themselves."We had to keep running the installation constantly to make sure it all worked. It was due to be there for three months and we had to stress-test it. Anyway, hearing those voices and whispers constantly made me start hallucinating other voices and whispers that weren't part of the composition. It was quite an intense experience."

Acting on impulse

Amazingly, all of the instrumental elements on the record (including an omnipresent low frequency hum, the shifting, pulsing chord cycle and a weaving organ-type sound) were developed using algorithmic musical programming, synthesised with inspiration from the odd acoustic properties of the building itself.

"I made different types of synthesis inspired by the acoustic properties of the prison. Most of it is phase modulation synthesis which I'm a fan of. There's also an instrument that works almost like a multichannel feedback unit, where you can have several octaves of processing and you can overrun each layer gradually. It goes from a delay-like feedback into a self-oscillating controlled feedback. It has organ-like qualities, but that similarity is very much all about overtones and how you use them. Since the vocal tracks were re-recorded from within the building, it works as a re-amp through the prison walls, further enhancing the spectral qualities of the building."

```
Server.local.options.numOutputBusChannels = 8 ;
s.boot

TempoClock.default.tempo = 140/60;

(
// acid bass Main Synth

SynthDef(\acid, {|freq=69, rq=0.2, amp=0.1, gate=1, out=0 atk=1, sus=1, rel=1, pan=0, lagti
    var env = EnvGen.kr(Env.asr(atk, sus, rel, -4), gate, doneAction:Done.freeSelf);
    var sig = Saw.ar(freq.lag(lagtime) * [0.98, 1, 1.01, 2, 0.5]);
    sig = Splay.ar(sig);
    sig = RLPF.ar(sig, env.linexp(0, 1, 60, 4500), rq);
    sig = Limiter.ar(sig.sanitize);
    sig = PanAz.ar(8, sig, LFNoise1.kr(0.05));
    Out.ar(out, sig.dup * env * amp);
}).add;
)


(//Acid bass Layer 1
PmonoArtic(
    \acid,
    \octave, 3, //+ Prand([ 0, 0, 0, 1 ], inf),
    \legato, Pwrand([ 1, 0.5 ], [ 0.2, 0.8 ], inf),
    \lagtime,  Pwrand([0.5, 1], [ 0.7, 0.3 ], inf),
```

Supercolliding

Originally developed and released in 1996 by James McCartney, SuperCollider is a cross-platform, real-time environment and programming language that is used for audio synthesis and algorithmically driven composition. It lay at the heart of Maria's approach to making spectrally originated synths and sounds. It was made open-source in 2002 and, since that time, whole fields of research have been done into developing and expanding the coding language to further manipulate sound in interesting and dynamic ways. It has three core components: *scsynth* is at the heart of SuperCollider's sound synthesis and processing, *sclang* is the language and sequencing section, that tells *scsynth* what to do (and when to do it) while the third element *scide* pertains to the editor and interface, which ultimately links *sclang* to *scsynth*. Since its launch, it has become one of the most widely-used platforms for serious sonic interrogators, artists and acoustic scientific study. More information on what it is and how to use it can be found at supercollider.github.io.

Maria expands on the benefits of building her own synthesis within the Supercollider framework. "As you get acquainted with this musical environment, you learn a lot about signal processing and how to incorporate algorithms and chance operations, you develop your own preferences when it comes to thresholds or controlled randomness, much like how you develop your own habits when patching a synthesiser. On the server I have my library of synths that I like using – then there's another paradigm on the client part of SuperCollider which is called 'patterns'. It's like an advanced sequencer. You can schedule events and build structures, as a way to sequence your pre-made synths. That's mainly how I've approached it."

While developing the multi-channel installation required its own space-specific consideration, the mix for the album release posed challenges, too. "It was a nightmare to mix it," Maria says. "As the installation was multi-floor, voices were coming from above and each side from the different rooms, so there were a lot of phase issues to deal with. The final stereo mix [for the] record was difficult. My first step was to compile the recordings into an adaption for surround sound that was presented at Fylkingen in Stockholm.

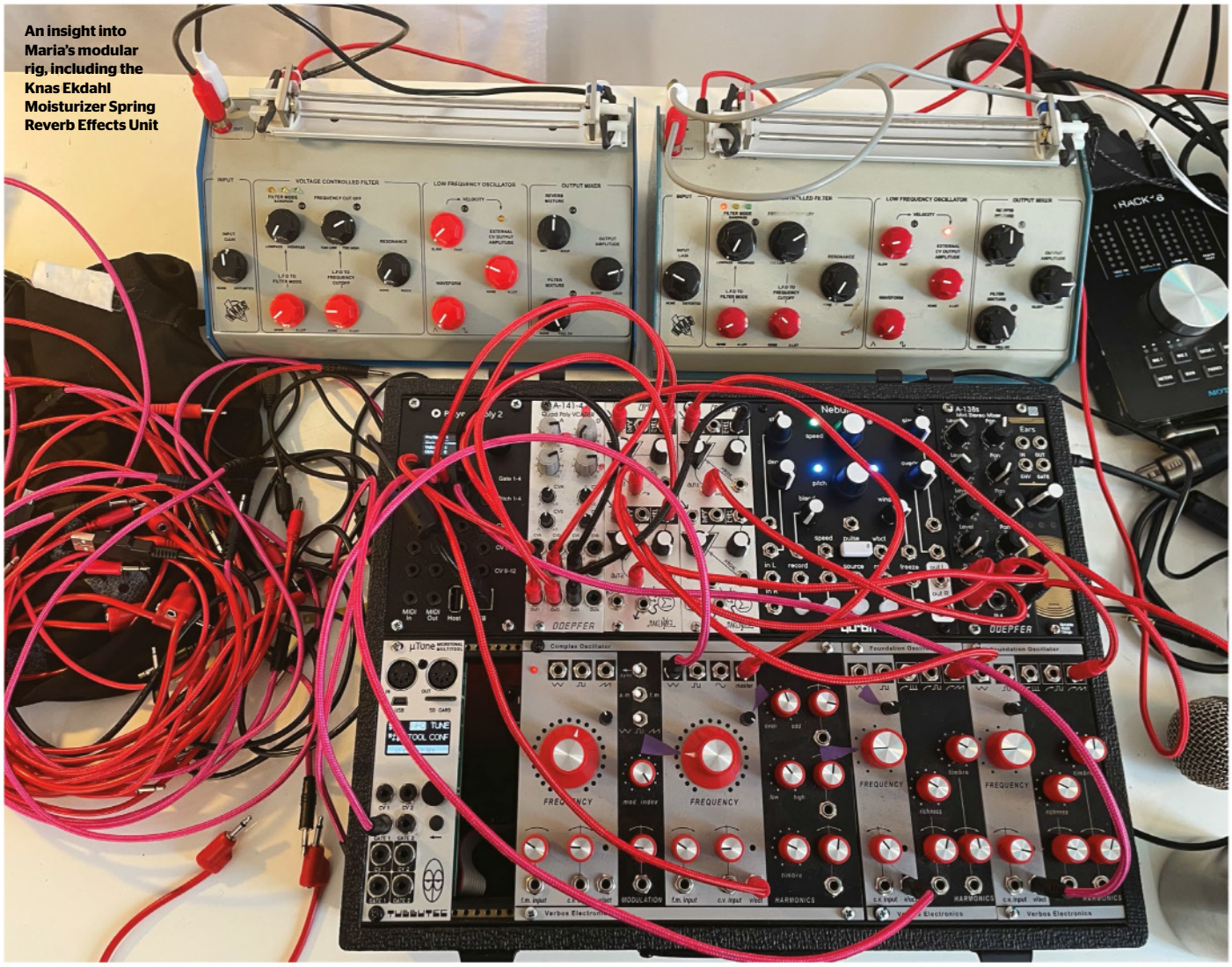
> "Artistic expression can become more intimate when there are more possibilities to adapt the tools of production"

"After this, I reworked the material a bit and mixed it to stereo, which of course had to involve a bit of compromise compared to the original spatial idea. In general I'm trying to make stereo adaptions of my multi-channel pieces, as spatial setups are kind of a rare thing to have access to.

"It's a different dimensionality of course, mixing in stereo," Maria continues. "As this composition is based on the spatial idea of a panoptic prison, a stereo system is a completely different listening experience. Immersion is a tricky thing. Either you're non-compromising and seek out the spaces that have these rare multichannel systems, but then you end up presenting your music mainly at institutions and within academic contexts, or you adapt the spatial idea to work in different systems, and I find great value in presenting my music in all kinds of venues."

We wonder what Maria's preferred mixing environment is? "I use Logic mainly, or Reaper," Horn answers. "*Panoptikon* was recorded in Reaper and then I migrated the files to Logic. I did the final mixing there. It's a blessing and a curse to get too comfortable with one environment early on. [In Logic] I don't have to think twice about what I'm doing."

At the outset of her journey into music production, Maria shunned the larger music production entities, in favour of more DIY



An insight into Maria's modular rig, including the Knas Ekdahl Moisturizer Spring Reverb Effects Unit

approaches. "There were a few years when I only wanted to use open-source programs. I had this idea of big tech companies streamlining the musical choices and compositional possibilities of the users by, for example, offering presets and genre-customised sample banks. I'm less restrictive these days, but I still believe that the artistic expression can become more personal and intimate when there's more possibilities to customise and adapt the tools of production."

We ask Maria how her journey into the world she now operates in first began. How did she start making music? "I started playing electric bass in punk and folk music bands in my old home town, Härnosand. At the age of 17, I was invited to participate in an evening course of electroacoustic composition. At that point I felt a little bit limited with the electric bass. Having studied the instrument in a jazz-oriented setting, there was a strong focus on virtuosity and being able to reproduce musical standards.

"Learning about electronic music was a bit of a new frontier in this sense. It was like an open field where I did not have to limit my imagination. I guess I got hooked. Since then I've been focused on making electronic and electroacoustic music."

Pattern forming

Maria explains that many of her closest collaborators are friends who operate within Stockholm's experimental music scene. "My main collaborators are people that I came across at the university, the EMS Electronic Music Studios or surrounding the venue Fylkingen. This venue is something of a national scene for experimental music in Stockholm. Unfortunately they don't have their space anymore [The Munich Brewery, a space they've had since 1986]. When I first moved to Stockholm, that's where I got in contact with new musical expressions.

"Together with American composer and organist Kali Malone, I co-founded the XKatedral label. We started out DIY-style, dubbing cassettes and setting up release concerts with our friends. We wanted to capture and expand on the interesting intersection between the local neo-classical, noise and electronic music scenes that were specific to the Stockholm scene. A few years in, we involved more people to further expand the scope of the label."

With an upcoming tour supporting experimental rockers Swans, we wonder if any of the *Panoptikon* tracks will find their way into Maria's live sets? "Some of it will be represented but re-adapted into live versions. Instead of the vocal recordings, I will use the vocal lines and sequence them using MIDI to my modular setup. But I'm not strict about playing the record, I will incorporate new ideas as well."

Maria wants to focus on developing the spectral approach further. "I have some new ideas for site-specific compositions where I want to expand the ideas of using the overtone-spectra as a starting point. I've also been composing four-handed organ pieces with my partner Mats. Rooms housing organs usually have interesting histories!" **cm**

For more information on Panoptikon *and find out tour dates, head to mariahorn.se*

HEAR MORE

Panoptikon
bit.ly/mwh_pan

Omnia Cittra Mortem
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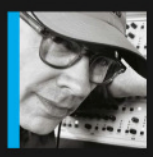
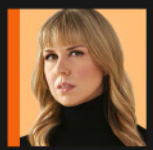


CM Tools Masterclass
with *Roland Schmidt*

38 Make a filter kick bass in BazilleCM

Get to know the best of our Plugin Suite with Roland's deep dives



Dance Music Masterclass
with *Sara Simms*

42 How to make rave-style techno

Acclaimed DJ Sara Simms blows her whistle and fuses '90s rave with techno

Mix Masterclass
with *Jon Musgrave*

46 Sidechain compression

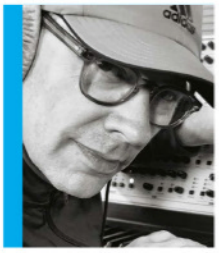
Jon Musgrave continues his quest to kick our mixing processes into shape

Download the accompanying video and the MIDI/audio files at **bit.ly/cm333downloads**

Make a filter kick bass with BazilleCM

Following our deep-dive into the oscillator section of BazilleCM, we follow the signal flow to create a filter-thumping bass sound

Roland Schmidt



Roland Schmidt is a professional programmer, sound designer and producer, who has worked in collaboration with a number of successful production teams over the last 25 years. He can also be found delivering regular and key-note lectures on the use of hardware/software synthesisers and production, at various higher educational institutions throughout the UK.

BazilleCM is a powerful beast. We've already established that the oscillator section hosts some amazing features, thanks to its association with phase distortion synthesis. This means that we can create some pretty complicated waveforms at the very beginning of our signal path. In true subtractive synthesis style, we have the option to feed both of the oscillators through the filter section, which offers us a wealth of sonic colour.

As many of us already know, the primary focus of the filter section is to filter out elements of the harmonic series that we don't want to hear. For example, we may choose to remove some of the upper harmonic brightness, contained within an oscillator's sound, leaving the lower frequencies to pass through, hence the description 'low pass' filter. The extent to which we apply this is governed by the position of the Cutoff Frequency pot. Sweeping this pot will provide the classic filter-sweep effect, particularly if coupled with the resonance control, which accentuates the point of cutoff.

BazilleCM is blessed with six filter types, which include four low-pass filters, a band-pass filter (which sweeps a band of frequencies either side of the cutoff point) and finally a high-pass filter, which let the high frequencies through, as an opposite to the low-pass filter. So now it's time to unleash the filter, and give our bass a good kick!

u-he
BazilleCM

A gem of our CM Plugin Suite, u-he's BazilleCM is our unique version of the company's patchable modular synthesiser, sequencer and mapping generator. It's a beast!
AU/VST 32-/64-bit

> Step by step

Make a filter kick bass with BazilleCM

1 > Let's begin by initialising a patch on BazilleCM. Open up your DAW and load up the plugin. Once the plugin window is open, move to the central display at the top of the window, click to reveal the dropdown menu, and select 'init' from the bottom. This will initialise the patch.

2 > As part of the initialisation, you'll also hear the FX Delay active in the signal. To make our adjustments clearer at this stage, go to the Delay section (bottom-right) and move the switch to the 'off' position. You can always switch it on again later, if so desired.

3 > Begin by creating a sound by heading to the Osc 1 section; locate the phase distortion (PD) pot, halfway down the oscillator section. Increase the pot value to its maximum (100). Don't forget, every time we make an adjustment, we can see an accurate value of adjustment by looking at the display at the top of the plugin window.

4 > Keeping our focus within the Osc 1 section, let's move down to the Fractalize controls; select Saw from the dropdown menu, and then adjust the Fractalize pot to a value of 0.08. This small but effective adjustment will act to substantially thicken up the texture of our chosen waveform.

5 > Next, we'll patch in the filter, but first disconnect the Osc 1 output. Click and hold the patch cable connected to Out 1, before releasing it on the Osc 1 output point, to disable the connection. Then create a new cable from Osc 1's output, inserting it into the filter input point. Connect the LP6 output point, and patch it back into Out 1.

6 > We'll introduce Osc 2 into the patch, but we'll also need to patch it to the filter. Create a patch cable from the Osc 2 output, and connect it to the second filter input. The inputs on the filter are not labelled, and effectively sum the signal, so you can use either the upper or lower input, as required.

Download the accompanying video and the MIDI/audio files at **bit.ly/cm333downloads**

> Step by step

Make a filter kick bass with BazilleCM *(continued)*

7 > Switching to the Osc 2 section, increase the Tune pot to a value of 7. Then move down the oscillator section and locate the PD pot, which we need to set to a value of 61. Finally decrease the output volume of Osc 2, setting it to 35. We've tuned Osc 2 to a 5th above Osc 1, as well as thickening its texture and decreasing its volume slightly.

8 > By default, Envelope 1 is controlling the amplitude or volume of our sound. This does not require a patch cable, as it's already modulating Out 1. We do however need to change the settings of the envelope; set the Attack fader to 0, Decay to 36, Sustain to 0, and the Release to 30.

9 > We can also use the output of Env 1, for the purposes of modulation. Create an output cable, which extends from Env 1 to the Filter cutoff modulation pot. These pots are not labelled, but can be found in the lower left-hand area of the filter section. Once connected, change the value of the pot to 116.

10 > The filter section will need to have some adjustments made, to counteract the modulation elements. We will begin by increasing the Gain pot to a value of 40. This has the effect of increasing the signal going into the filter circuit, albeit virtually. You should hear a radical increase in volume.

11 > Keeping within the filter section, set the Cutoff pot to a value of 0, before setting the Resonance pot to a value of 49. Despite the huge decrease in the cutoff value, the modulation will now come into effect in a huge way, giving us the true kick bass sound.

12 > Finally, we'll create another modulating output from Env 1. Create a second patch cable, from the output of Env 1, and assign it to the PD modulation pot, just left of the main PD pot in the Osc 2 section. Once connected, set the pot to 60. This now means that we're modulating the phase distortion element of OSC 2, as well as the filter.

PRO TIP

FILTER POLES AND dB CURVES

You may have noticed, that in the creation of this patch, we specifically used the filter described as LP6. This is because we wanted a particularly aggressive sound, but what does the filter name actually refer to? The LP part of the description refers to 'low pass', which lets the lower frequency content pass through unscathed. And the number? Filters often have a number attached, such as 24, 18, 12, or 6. Each of these numbers is a value which reflects dB per octave, which describes the slope of the filter. Your choice of filter is dictated by the sort of sound you want to create. Within BazilleCM, there is plenty of capacity to explore the different filter types, as they all sound different. To make matters even more confusing, there is another term which is similarly employed to describe these filter types, namely 4-pole, 3-pole, 2-pole and 1-pole, hence a 24dB filter is often described as a 4-pole filter. This element colours the sound considerably, and programmers have an idea of which filter to use for a required sound. We're using the Low Pass 6dB filter in this patch, with its relatively harsh sound, ideal for a filter kick bass.

Lindstrom sits back and enjoys his filter kick bass work in action

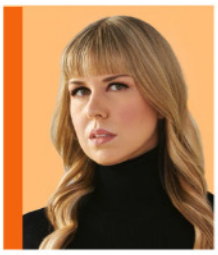
RECOMMENDED LISTENING

LINDSTRØM
RÅ-ÅKŌ-ST
On this opening cut from the album *Smalhans*, you can hear the kick bass effect right from the very first note of the track.
bit.ly/lndstrm_r

CALVIN HARRIS
HOW DEEP IS YOUR LOVE
This classic track from Calvin Harris also adopts a kick bass principle, but in this example, the bass is squarewave-shaped, with less filter involvement.
bit.ly/ch_hdiyl

Download the accompanying video and the MIDI/audio files at **bit.ly/cm333downloads**

Sara Simms



Sara Simms is an acclaimed DJ and producer from Toronto who's played some of the world's most prestigious venues and festivals alongside artists like Carl Craig, Victor Ruiz and more. She owns her own record label, Simmetry Sounds which releases cutting-edge techno and house tracks crafted by the next generation of electronic music producers

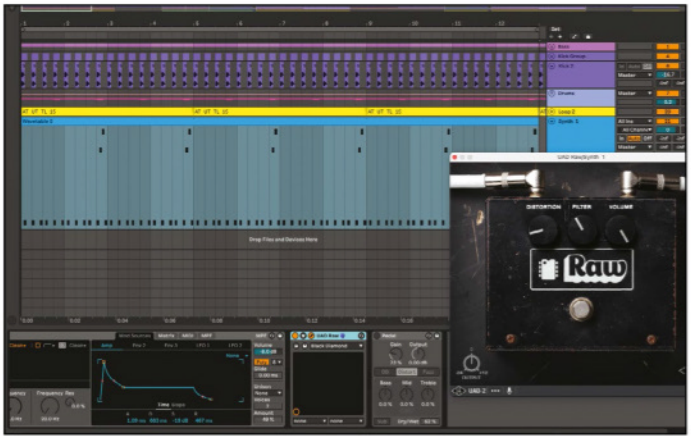
How to make a rave-style techno track

Prepare to get fists in the air as we learn the tools, tricks and tempos for nailing the basics of rave-ready techno

A popular sound in techno today is a hybrid of modern production techniques and the rave sound of the '90s. In this merging, faster tempos, detuned synths, melodies and acid basslines are fused with modern techno sounds. A multitude of music production techniques were born in the rave scene in the late 1980s and early 1990s, the emphasis for producers from this era was firmly on inventiveness and experimentation. Plugins and software synths can allow us to creatively recreate sounds that bear resemblance to sounds from the rave era. In this month's tutorial, we'll look at how to create rave-style techno, including building an atmospheric synth and trance synth using Ableton's Operator. To lay the foundations for your track, program in a kick drum with a strong transient, and sub bass. Use closed and open high hats and a clap from a 909 drum kit (NI's Battery, or D16's Drumazon are good choices). Rolling basslines come next, and then it's time to start thinking synths...

> Step by step

1. Building an atmospheric rave-style techno track

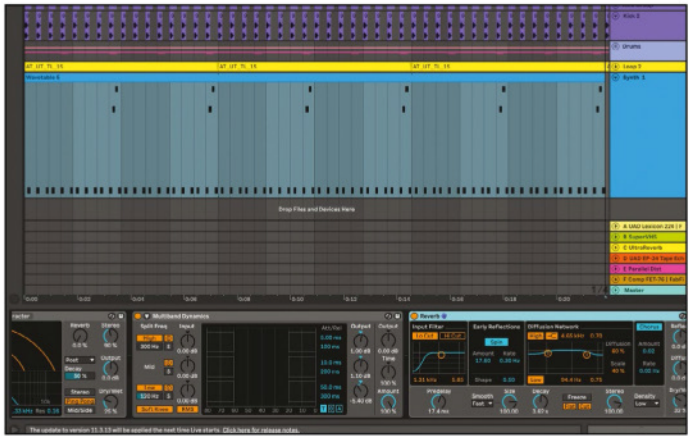


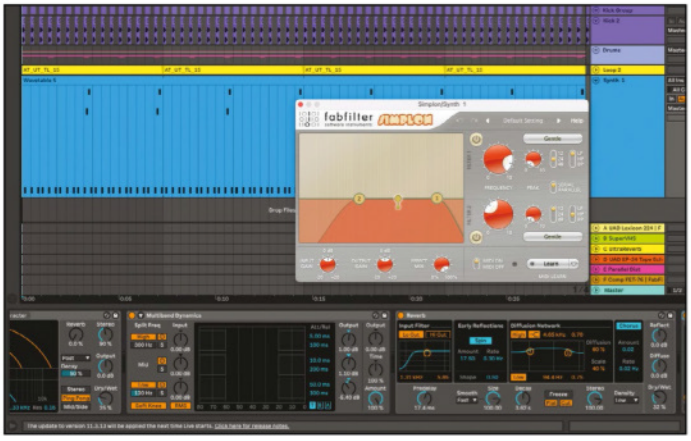
1 > Open Ableton Live and an instance of Operator. In Osc 1, choose Sub 1 and set Osc position to 21% and Fold to 20%. On Osc 2, choose Harmonic Series and set it to 71%. In the Filter section, set the Low Pass Filter's Resonance to 0 and Frequency to 20Hz. Drag the Frequency knob to the Matrix to assign it to the Amp with a value of 52.

2 > In the Mod Sources, set the Amp to 1.09ms, Decay to 683ms, Sustain to -19dB and Release to 467ms. Add in a simple melodic pattern, ideally one that repeats three times with a variation in the fourth bar. Add an aggressive distortion (We're using Universal Audio's Raw) with the filter around 11 o'clock, distortion low and volume high.

3 > Next, we further dirty the sound with one of our favourite saturators, FabFilter's Saturn 2. Setting the crossover point to 300Hz, and bring Band 1 to around -3dB. We set the Band 2 level to +8dB, and the Band 1 Drive to 65%. Set Band 1 Level to -2.73dB. Using an EQ, roll off the low end below 150dB and cut all above 5kHz.

4 > Add Ableton's Chorus-Ensemble to give the sound movement. Set the Rate to 0.26Hz, Feedback to 47% and Dry/Wet to 50%. Add Ableton's Echo with a Dotted ⅛ Echo, Feedback at 50%, and Bandpass Filter focused between 600 and 1.5kHz with a small amount of Resonance. Turn on Ping Pong Delay and set the Dry/Wet to 25%.

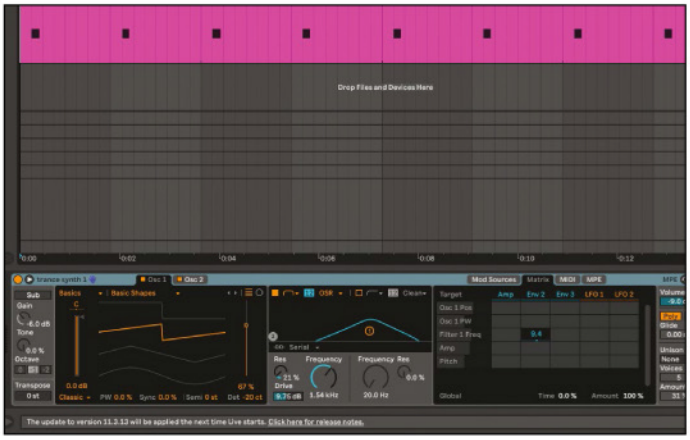




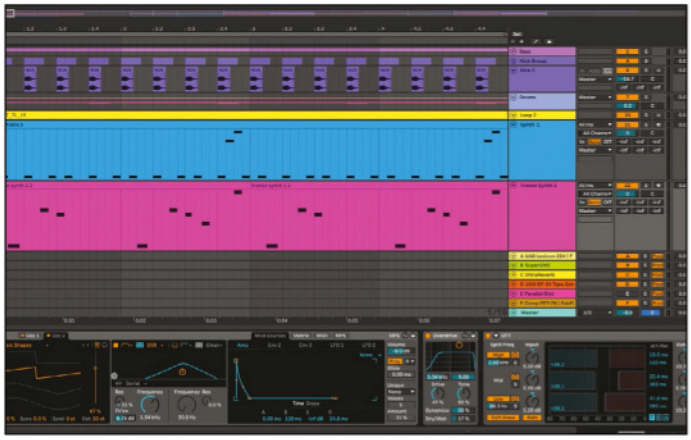
5 > Add Ableton's stock Multiband Dynamics to process the three frequency bands by applying compression. Set the Output of the High Band to 1dB, Output of the Mid to 1.1dB and Low to -5.4dB. Set the Amount to 100%. Add Ableton's Reverb and set the Predelay to 17.4ms and Decay to 3.62s. Set the Dry/Wet to 32%.

6 > Lastly, add or Ableton's Auto Filter (or a filter like FabFilter's Simplon). Cut off the low frequencies below 150Hz, and cut high frequencies above 20kHz. Highlight each of the plugins and right click to save them in a Group. Save the chain in the User Library in Ableton by clicking the disk in the upper right hand, give it a unique name.

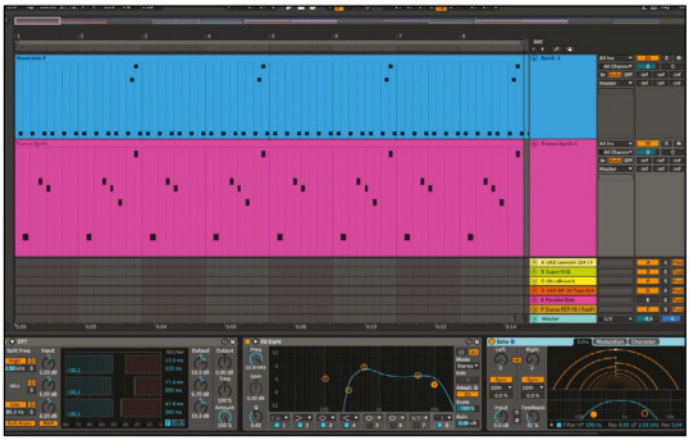
> Step by step 2. Trance synth



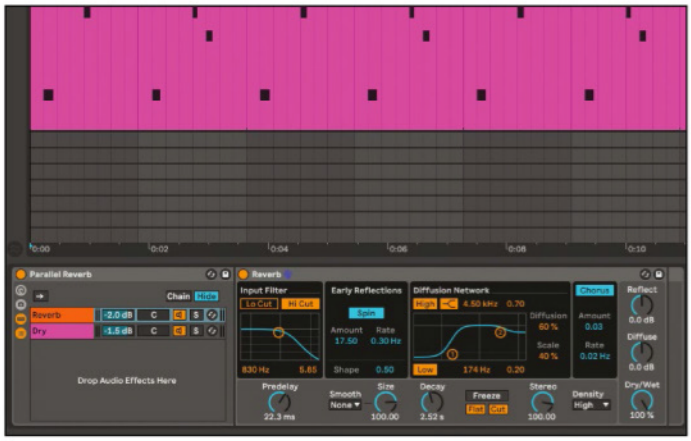
1 > Create a MIDI track and add Operator. In Osc 1 and Osc 2, choose Basics Shapes and set the Osc position to 67%, then Detune to -20. In the Filter section, select the Bandpass filter and set Resonance to 21%, Frequency to 1.54kHz and Drive to 9.75. Click on the Frequency and drag it to the Matrix on Env 2 with a value of 9.4.



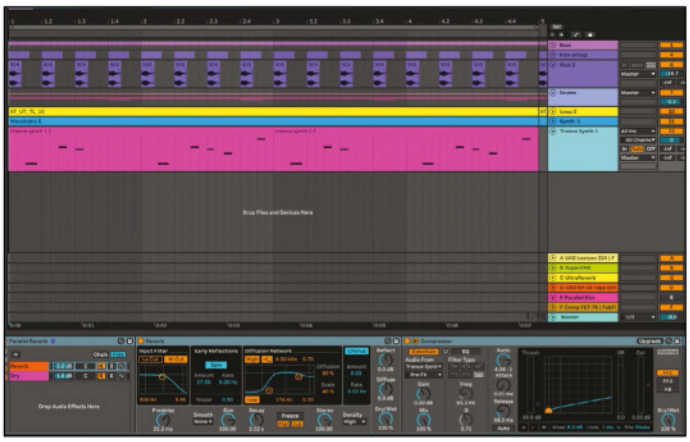
2 > In the Mod Sources, set the Attack to 0, Decay to 129ms, Sustain to -inf dB and Release to 24.8ms. Create a catchy one-bar melody. Add aggression by adding Overdrive, focused around 3.33kHz. Set the Drive to 47%, and Dry/Wet to 17%. Add OTT, and adjust the output of each band to taste. Set the amount to 100%.



3 > Add an EQ and cut the low end below 270Hz, boost the mids around 500Hz and roll off the top end above 10.9kHz. Add movement to sound by adding an echo and selecting a 1/16th note. Set the Feedback to 50%, Dry/Wet to 22% and turn on Ping Pong. Use a bandpass filter to focus the echo around the midrange.



4 > Create a parallel reverb chain that allows you to mix the processed signal with the dry signal. Add a reverb, highlight it and select Group. Turn on the Hi Cut and roll off frequencies above 800Hz. Set the Predelay to 22ms, Decay to 2.5s and Dry/Wet to 100%. Right click inside the Group to select 'Create Chain' to create a dry chain.



5 > Add Ableton's Compressor to the parallel reverb chain, and sidechain the audio from the Trance Synth, and select Pre FX. Set the ratio to 4.34, attack to 0.01ms, and fast Release around 60ms. Set the threshold to -40dB, and set the Dry/Wet at 100%. This compressor gives the synth a tighter sound, with less reverb tail.

6 > Add a second compressor after the parallel reverb chain and sidechain the compressor to the kick in your track. Set the Ratio to around 4:1, the Attack to 0.01 ms and Release to 13.6 ms. Set the threshold to taste, around -10dB and the Dry/Wet at 100%. You could also use a dedicated plugin like Cableguys Kickstart 2 for this.

Try these plugins to help shape your synth stab to perfection. Clockwise from top left: Reaktor Lazerbass, FabFilter Saturn 2, Arturia Rev Intensity, Soundtoys Echo Boy

PRO TIP

SYNTH STABS

A synth stab adds character to a rave style techno track, and brings a modern edge to the production. A VST library like Native Instruments' Reaktor or Spitfire LABS is a great place to look. For our example, we selected Reaktor's Lazerbass, but scroll through what's on offer and find an instrument you like to make it your own.

To get started with shaping your 'stab', create a MIDI pattern and place your sound where there's some space. In order to start processing, add an EQ and roll off the low frequencies below 100Hz.

Add some saturation to the stab (we used FabFilter's Saturn 2 but your DAW may have options). Try setting the drive at a low value, turning up any crunch or overdrive and setting dry/wet to 100%. Add a 1/16 note delay (try something like Soundtoys EchoBoy for this), and set the mix to a low value. For thickness, use a Neve preamp-style effect like Kush Audio's Omega N. Shape the result using a vocoder, or even just a band-pass filter.

Lastly, add space using a reverb. Choose a preset with a predelay of about 15ms, a large room size, and set dry/wet to a low value.

RECOMMENDED LISTENING

RAMIRO LOPEZ AND SPARTAQUE
CÁLICO ELECTRÓNICO
This track incorporates harder-sounding rave-style synths, and a breakdown that's very '90s-inspired.
bit.ly/rls_ce

ELI BROWN
SEND HELP
This track combines a lead synth that's reminiscent of the rave era and modern techno elements.
bit.ly/eb_sh

Download the accompanying video and the MIDI/audio files at **bit.ly/cm333downloads**

Jon Musgrave

Jon is a London-based platinum award-winning mixer, producer, composer and club remixer with a diverse CV that spans dance, pop, rock and music for media. He's also a long-term contributor to *Computer Music*. Jon usually handles final mixdowns, which is why we've got him to share some of his pearls of mixing wisdom here

Sidechain compression

Sidechain compression is an excellent mixing tool and in this tutorial we pick apart this essential process

Most commonly associated with the kick-driven, audible pumping effect that's become popular in club music, sidechain compression can be a very useful tool that adds energy and space to individual sounds, submixes and mixes while keeping things punchy. It can be used in many different ways and so we'll be splitting the topic across more than one tutorial.

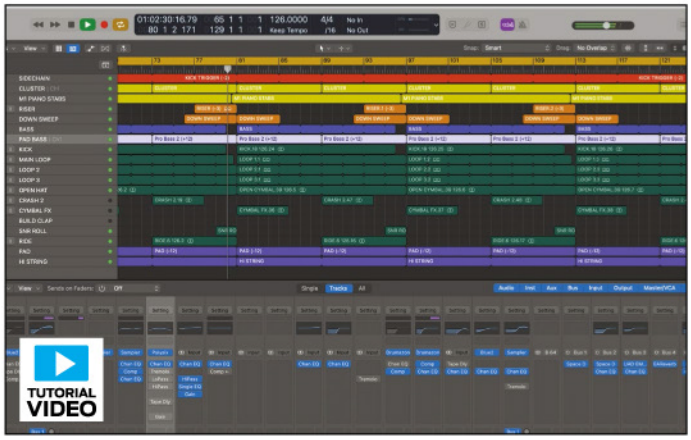
From a technical perspective, sidechain compression typically means the use of an external input to control the compressor, and this can be a sound that's part of the mix or a specially created key. Sidechaining has its roots in analogue processing and a small number of hardware compressors that incorporate an external sidechain input. This extra flexibility allows all manner of outcomes including frequency-specific compression if you parallel and filter your source signal. It also allows the more creatively minded to achieve more extreme pumping effects that are popular in house, EDM, trance and beyond.

Many DAWs include immense routing flexibility, with multiple busses and precise automation. This allows us to deliver highly finessed and creative sidechain compression treatments. Nevertheless, the core technique remains basically the same.

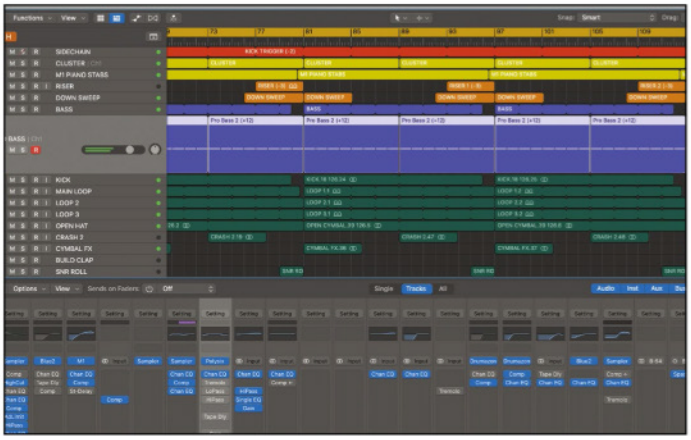
In this tutorial we'll be focusing on the classic club effect, showing you various ways to get set up and how we can use sidechain compression. We'll even throw in an alternative way to get a similar effect – check out the Pro Tip.

> Step by step

Setting up sidechain compression



1 > There are various ways we can set up sidechain compression, so let's start with the most basic. We have a simple house track with a number of bass, pad, string and loop components in addition to the main beats – kick, hi-hats, crash and so on – but you can apply the same technique to many styles.



2 > Our main bassline does not play at the same time as the kick, so we'll leave the bassline alone. However, we do have a low bass pad sound that helps fill out the low mid frequencies. This is hard to balance and not particularly dynamic but sidechain compression can change that. Pad sounds in general are good targets for sidechaining.

3 > To get started, select a compressor with external sidechain options and insert it across the sound you want to process (pad bass for us). Now assign the kick drum track as the external sidechain source. To get a pumping sound from the compressor, you need to focus on the attack and release times, and then adjust the threshold to taste.

4 > Set a very fast release (ours is 5ms), fast attack (ours is 20ms) and medium/high ratio (6:1). Now adjust the threshold until you hear quite a strong effect – your gain reduction meter should be pretty active. If you now tweak the release time, you can tailor the shape of the compression. We've set ours to 50ms for a moderate effect.

5 > This technique works fine but the sidechain effect stops whenever the source track stops. This may be what you want but can also be a bit restricting. With that in mind, it can be easier to set up a bespoke sidechain track. Here we've simply duplicated our drum track, edited it to taste and made sure it's not routed into the mix.

6 > A further (and more flexible) option is to use a bus as your sidechain source, assuming your DAW allows this. The advantage is that you can create a sidechain signal blended from multiple tracks and also adjust and automate the level of that signal. Here we've automated our bus send to control how much compression occurs.

> Step by step

Setting up sidechain compression *(continued)*

7 > Sticking with the subject of amount, we normally achieve this by adjusting the threshold on our compressor. However, if your compressor has continuous ratio adjustment from 1:1 upwards, adjusting the ratio often works better. So, for this, we've controlled the amount of compression by automating our ratio.

8 > So far we've only looked at sidechaining one track. To create a more pronounced effect, we really need to use the technique across multiple tracks. There are various ways to do this and we can simply replicate the basic technique. This is flexible, as each track can be tailored, but it's also a bit fiddly.

9 > Here we've extended our sidechain compression targets to include a pad and also a high-string part. Using our basic technique, each track gets its own sidechained compressor and they're all controlled from the same bespoke track. The flexibility lets us have quite a deep effect on the high string and more moderate effect on the pads.

10 > Multiple tracks can also be processed collectively and this can create a more uniform effect. To do this, all you have to do is send all the desired tracks to a fresh bus, creating a submix. Then set up a compressor on the submix and route your desired signal to the compressor sidechain input.

11 > Here we've bussed together a number of components including some of the beat loops and some effects to create a submix that we then process with one sidechain compressor. You can easily roll out this method across a few submixes to create a sidechain effect that works for most aspects of your mix.

12 > Finally, don't ignore your reverbs. Sidechaining can work very well with longer reverbs, particularly if they're on auxiliary returns, and it lets you create some sidechain-style pumping on the effect return without influencing the source sound. Set up a compressor on the return and apply your chosen sidechain method.

PRO TIPS

ALTERNATIVES

Using a compressor is just one option for making a pumping sidechain effect. Another is a plugin called Nicky Romero Kickstart 2 from Cableguys. This offers envelope tailoring plus various tempo-synced timing options, as well as MIDI triggering. Similar is the great Duck from Devious Machines, which lets you create pretty complex bespoke curves that go well beyond the regular sidechain compression effects. Ableton Live users should also try the Max for Live device Duck Buddy. These plugins use tailored and triggered volume envelopes, as per the effect they're trying to create. If you simply want a tempo-synced pumping volume effect, a tempo-synced modulation can achieve that. Plugins like Xfer's LFO Tool fit the bill, but you might also find that a LFO-controlled volume plugin like an autopan might have enough flexibility to do this. Logic's Tremolo plugin is a perfect example. If you set the Phase to 0 and choose a quarter note Rate, you can use the Symmetry and Smoothing options to tweak the shape and the Depth setting as an amount.

Your options for pumping sidechain effects don't end with a compressor

Kylie's *Padam Padam* alone could merit its own mixing masterclass

RECOMMENDED LISTENING

TOGETHER
SO MUCH LOVE TO GIVE
Classic full-on sidechaining makes this French house classic super memorable
bit.ly/tg_smltg

KYLIE MINOGUE
PADAM PADAM
Clever use of selective sidechaining adds flavour to specific sections of the track
bit.ly/km_pdm

SEDIBUS

Detaching from The Orb mothership, Sedibus is the latest project of core figurehead Alex Paterson, accompanied by former Orb-member and key creative collaborator Andy Falconer. With a studio shorthand and a shared passion for flourishing ambient textures, the pair's second record *SETI* gazes skyward for its inspiration. We spoke to Andy to learn more...

> As one of those foundational outfits that defined so much of what we now take as read in the electronic domain, The Orb needs little introduction. Back in 1988, their beloved debut, *The Orb's Adventures Beyond the Ultraworld* set a new benchmark for evocative ambient and dub.

While subsequent records and line-up changes gave us further mind-expanding sonics (namely 1992's *U.F.Orb* and 1995's *Orbus Terrarum*), it's that perfectly titled debut that remains their high watermark. Beyond The Orb's creative lynchpin Alex Paterson, former studio engineer Andy Falconer became a key member of the initial lineup who assembled that debut. Decades on, and following a 20+ year family-focussed sabbatical, Falconer and Paterson reunited once again, this time under the new guise of Sedibus. Their immersive 2021 debut *The Heavens* was a thoroughly captivating listen. Impressive sales and critical giddiness has led the pair to expand on the Sedibus universe with follow-up, *SETI*.

"There was never a multiple-album deal or anything, [Sedibus] was just spurred by the idea of getting together again and making a record," Andy explains, when asked to take us back to the beginning of the Sedibus project. "Obviously between me and Alex last properly working together in the studio, I'd done some remixing for The Orb and we'd stayed in touch over the years. But it had been over 30 years since we'd

Never Trust
an
Orb

properly done something together creatively. It was at Space Mountain festival in Spain where we met up, and Alex said, 'Hey, how do you fancy making a record together again?' So, it was just a case of bouncing ideas backwards and forwards, to see what would happen. It very quickly clicked. There was really no problem creatively."

This ease of working resulted in that glorious debut, with the record's most divine moments (such as the 19-minute, space-angled epic *Unknowable*) pushing the pair to reconvene for album 2. "Cooking Vinyl loved the first album, and definitely wanted to get behind it. Luckily enough, people who heard it liked it as well. Of course, then it was obvious to do a follow-up to it. The follow-up was done rather quickly, and it's been sitting on the shelf for a year and a half. It's just been waiting for the right time period in Alex's activities – he's a very busy guy and runs his own label, Orbscure Recordings." It's worth noting here that The Orb is still very much alive, currently co-fronted by Alex and Michael Rendall.

The outer reaches

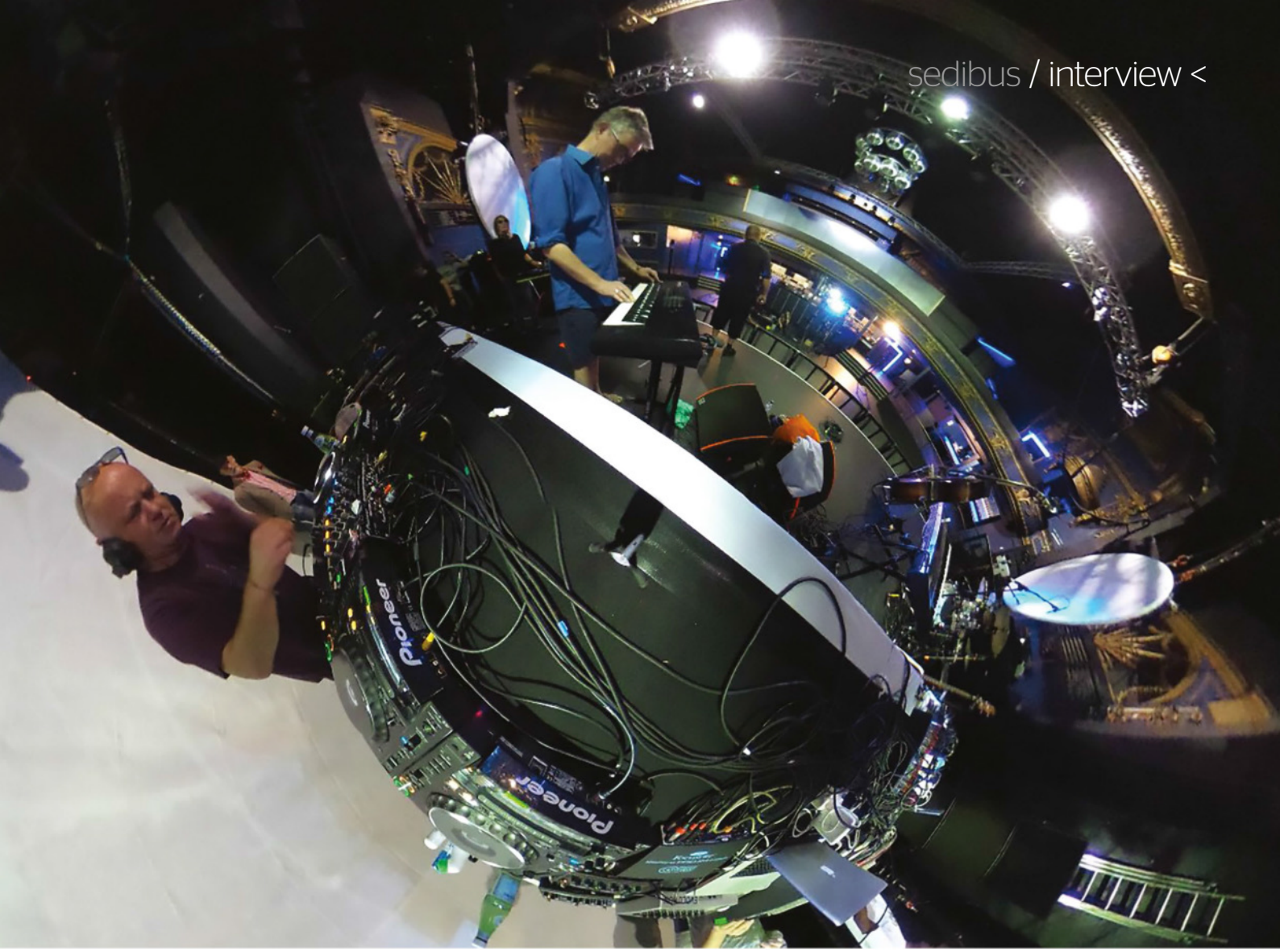
While the first record swam in dense synthetic currents, and told a gradually evolving sonic story, *SETI* incorporates a wider textural palette that includes a range of acoustic instruments. As Andy says, "it would have been very easy and very tempting to just continue in exactly the same vein as *The Heavens*, but the only plan really was to try and develop the idea a little bit further. One of the things we thought was to look at using more acoustic instruments, alongside the synthetic elements. I was the first off the mark at the beginning of Sedibus in sending Alex the idea for a track that was deliberately a little bit provocative. Alex replied saying that he 'fucking hated' it. It was really kind of funky – so I looked at the arrangement again, and that turned into *Unknowable*. When it came to do the second record, it was Alex that was first off the mark trying to shake things up."

> "I sent Alex an idea that was deliberately a little bit provocative. He said he 'fucking hated' it!"

Instructing Andy to tune in to his popular WNBC radio show, Alex transmitted a three-hour set of new and original ideas that Andy could develop. "He said 'get ready to record.' It must have been a pretty strange show for anyone who was listening, because he was basically just transmitting three hours worth of music to me, with loops, textures, musical parts and other bits. I just got three hours in one go of potential ideas from Alex. Then we spoke on the phone afterwards, and talked through some of the ideas. The first thing I actually did was sit down and cut up three hours worth of potential samples and parts. Made musical 'bits' and turned loops into rhythms. I put some stuff into Steinberg's Padshop and created new material with it."

Interestingly, both Andy and Alex kept the limitations of vinyl quite firmly in mind when assembling the record, prioritising it as the listening experience of choice. "The album was pretty much recorded in the order you hear it on the record. We were both thinking primarily about people listening to it on vinyl, and worked to the time constraints of the format. We started sharing ideas via email, and the album developed from there."

Andy's home studio hub, described in brief: "Cubase is my primary DAW, my desk interface is the Behringer X-Touch and I use the UA Apollo Twin for monitoring"



The speed of light

With the mention of Steinberg's Padshop, we assume that Cubase is Andy's primary working environment? "Yes," he confirms. "In fact, everything is software-oriented these days. I come from the old-school. The analogue tape and mixing desk-era. Whereas then everything was largely outboard, everything is virtual now. So, Cubase is my primary DAW. I'm still on version 10.5. My desk interface is the Behringer X-Touch, and what I use for monitoring is the Universal Audio Apollo Twin. Also, [UA's] software emulations are really, really good."

Synth and sample-wise, Andy explains that Native Instruments Kontakt is a big workhorse, and he elaborates further on his use of Steinberg's Padshop: "I like it a lot because you can just pull in your own bits and pieces. When we made *The Heavens* there was a moment where Alex sent me a YouTube video link to a particular ambient piece, and said 'I really like what this guy is doing, can you take some inspiration from that?' So I just took the entire 12-minute track out of YouTube, banged it into Padshop, played around with it and made a completely new soundscape and texture from it. This became the basis of one of the new tracks."

Steinberg Padshop is one of Andy's main go-tos for chopping up sonic elements and creating new samples

We ask Andy what, beyond Cubase, he couldn't work without. "It probably doesn't fall exactly into the area you're thinking of, but the thing that I've always loved in the studios when I started my career, is a pair of Generic 1030A self-powered monitors. I'm old school, I like the older versions. I love them as near-field monitors. I've always tried to work with them in my own studio. That's probably been the constant, or the most obvious constant in my workflow."

With his decades of studio experience, we wonder if Andy has any misgivings about the software-saturated world we (and he) now operate in, when compared to the practical skills required to use gear in the old days? "Well, I like the amount of automation that's open to you [with software]," he says. "Having everything based in software – whether it's filters, EQs,

> "We always trust that one of us will come up with the goods. We're both inquisitive and like to try new things"

effects-sends; you have so much potential for playing around. The main thing with it all being software-based is that it's an awful lot quicker to do anything. I remember back in the days when you'd have to sit in front of a Studer tape machine with a razor blade and cut everything up to make the arrangement. You'd have tape around your neck and it was so messy. These days you can just drag and drop with your mouse, it's tremendously quicker. The time-stretching capabilities of just pressing a button and getting something to loop immediately into the key that you want is fantastic. I remember the Akai S1000 when that ability first came in, it was a revelation then, but you were still playing around with the algorithm. Nothing was exact."

Andy continues: "Now you have way more creative possibilities open to you because you can try things out quicker, you're not committing yourself to one thing. We can just do it. It did have a certain charm – that old way – trying out different amps and putting them in different rooms etc, sending signals out, miking them up, bringing them back. Creating rooms and different sound effects, now you do it all virtually. You gain on the one hand, and lose a little bit on the other."

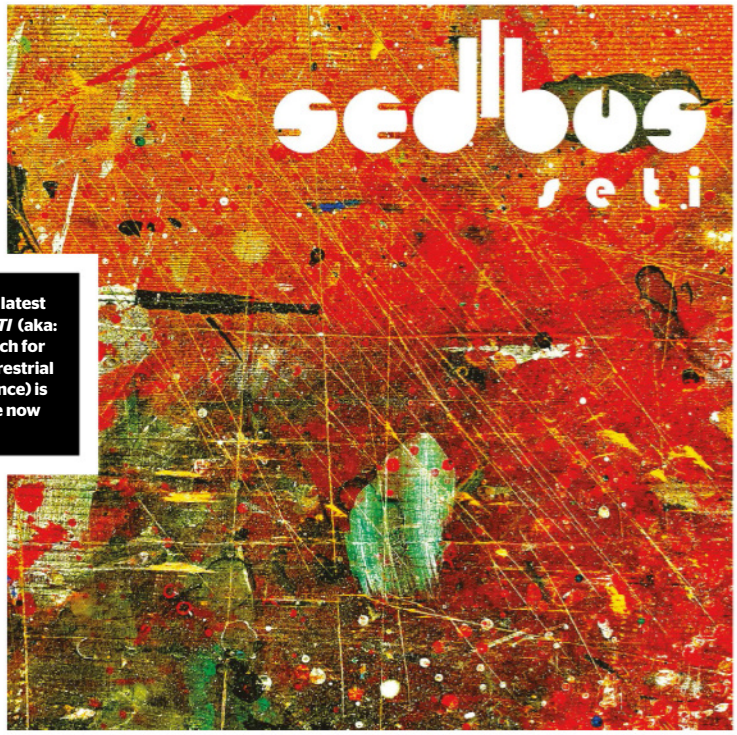
With his fond memories of that messily charming analogue world, it's no surprise that Andy still regularly uses tape distortion, albeit in software form: "I love virtual tape emulation. Going back to UAD, I really like their Studer A800 software. The kind of harmonic distortion you get from that is so authentic. I use it on an awful lot of things. Even if you're not really aware of it. I do actually like playing with a lot of distortion. I sometimes bring in an old favourite of mine, the good old Quadrafuzz (a Cubase stock distortion). It gets used quite a lot. So, I guess I'm adding in some of that analogue flavour when I can."

Beyond the Ultraworld

In his home studio, currently set-up in Portugal, Andy developed Alex's initial ideas and collaborated long-distance. "We always trust that one of us will come up with the goods. I know Alex's musical headspace very well and I think it's the same with him for me. We're both very inquisitive and like to try out new and different things. We're not shy of following things and seeing where they go. I think it's that shared curiosity that leads us to discovery, and we blindly know where the other person is at. When you've known each other for so many decades, there is a certain amount of unspoken communication where you know where the other person is going with something almost before it happens. It's one of the great, magical things about making music with other people."

We wonder how the division of labour is decided. Andy stresses that at no point do they

Andy admits that getting analogue tape emulation and saturation is far easier with software than in the old days



Sedibus' latest opus, *SETI* (aka: The Search for Extraterrestrial Intelligence) is available now

narrow down who does what. "There's never any sort of apportioning work where we decide who does what," he clarifies. "It varies definitely from track to track. It's also how you quantify the input as well. Sometimes, Alex has more of a guiding hand, sometimes it's more of a case of actually coming up with the physical bits and parts that are being used, other times it's the other way round. There's no hard and fast rule. I always look at the records and think they definitely are very much a 50/50 collaboration, regardless of who exactly played how many notes on what tune! When it comes to the final mixes, that's down to me. At the same time, there's tons of input from Alex, and I'm sending stuff over to him the whole time."

The record's thematic concern with extraterrestrial intelligence is borne out by its title, *SETI*, aka: The Search for Extraterrestrial Intelligence, a collective term for any and all organisations looking heavenward in the hopes of communicating with non-Earth based intelligences. It primarily refers to the SETI Institute, an American nonprofit dedicated to searching for little green men and everything else beyond.

Back to Sedibus, we're intrigued as to why the pair aligned their creative crosshairs onto the search for ET? "We both have a shared love of space exploration," says Andy. "It was way more exciting when we were kids. Very early on, Alex came up with the idea of calling it *SETI*. I basically ran with that and started looking for things that married up with that core idea. I came up with quite a few of the vocal samples. One difference between this and the first album, was that the vocal parts here came mainly from me, while the first album had mainly Alex's. I found a great search engine where you put in what you're looking for, and it'll trawl the internet and provide a huge amount of usable media pertaining to a topic. That's where I got things like obscure US podcast audio, where they discuss whether Jesus was an alien and that kind of thing. When it came to the B-side of the album, the first track was done and the second track was really sort of a recycling of stems that hadn't been used on the first track, becoming a part two. That's when we hit upon having *SETI* as a three-part cycle."

Across *SETI*'s runtime, consciousness-lifting sonics whirl into hypnotic micro-galaxies of sound. Spanning five tracks, the record opens with the acoustically-tinged *Paradise* and *Purgatory*, and culminates in the three-part *SETI* cycle. It's this extraordinary suite that Andy cites as his favourite part of the record. "I think I'd have to say *SETI – Part 2*, specifically," Andy says. "I like how it starts off very electronically. It has all these bleeps and synthetic textures, but then the whole thing morphs into an acoustic thing, which ends up sounding like a real band could stand up and play it. I like how it nicely merges from one to the other."

> "Whether a third record would continue being ambient, or whether we'd go dub, it's hard to say"

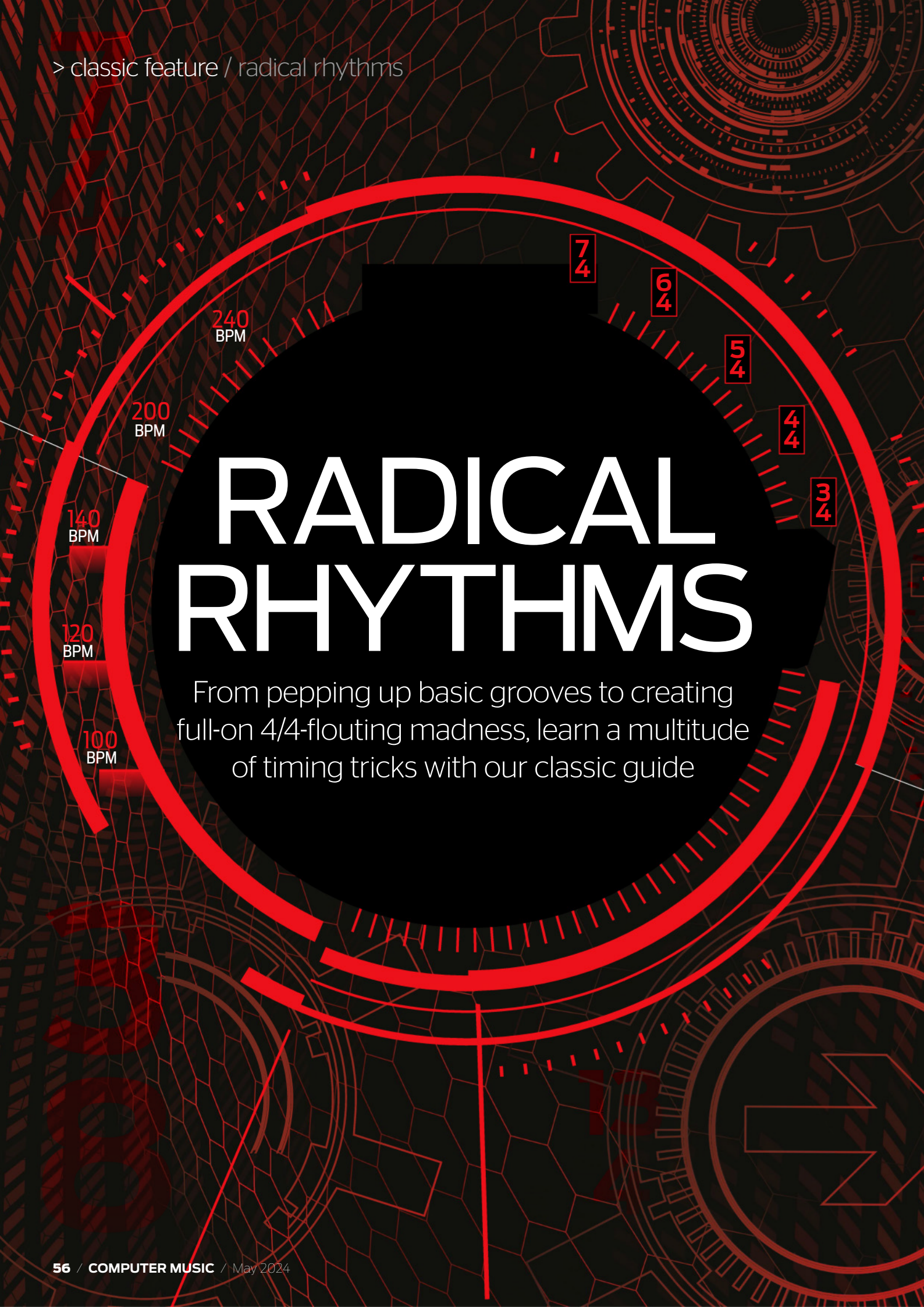
Prepare for launch

With *SETI* now released, we wonder if Sedibus is set to continue venturing into aural pastures new, or whether *SETI* marks the project's resolution? "Well, there's always the possibility of doing more," muses Andy, "but at the end of the day, we've made these records based on our creative drives, and we made what we wanted. From one point of view it's an egotistical thing when we're making it – we're focused on pleasing ourselves. But, of course, you're not working in a vacuum the whole time. When it's finished, someone will invest money and make vinyl and spend time promoting it. It's got to sell at the end of the day. As long as there's a market for it then there's the chance that we'll make another one. One of the big things is that we had a lot of fun getting together making these two records. It's been natural and easy."

Future live dates for Sedibus haven't been decided at time of writing, with Alex's continuing responsibilities as The Orb's linchpin taking up a great deal of his time. But as far as Andy is concerned, the pair could hit the stage relatively easily: "There's been noises made with regard to that, and I know that we're looking at ways of doing it. At the moment, I don't have any firm dates. We can do stuff pretty much at the last minute. Last year we played a Sedibus gig at Alex's Psychedelic Orbscure Ambience Festival. I put the live set together in Portugal and travelled over to England. The plan was that we were going to have a run-through beforehand, but that didn't happen, so we just turned up on the night, set up and played the album together for the first time without doing a rehearsal."

Andy stresses that, personally speaking, he'd be delighted to collaborate with Alex on further Sedibus projects: "There's definitely lots of new things we'd like to explore together. When we made *The Heavens*, the concept at the start was to make a record that reflected and referenced our shared roots – going back to *Ultraworld*, but it also showed how far we've come from that. With *SETI*, the feeling was 'how can we continue on from this development?', 'what do we want to explore next?'. With this record, it was primarily the acoustic textures that we brought in a lot more. Whether a third record would continue being ambient, or whether we'd go dub, it's hard to say. But it would incorporate elements we both enjoy. It all depends on how we're feeling when we sit down and decide what to do." **cm**

SETI is available now via sedibus.tmstor.es or theorb.bandcamp.com/album/seti
For further information on the Sedibus project, and possible live dates, check theorb.com



RADICAL RHYTHMS

From pepping up basic grooves to creating full-on 4/4-flouting madness, learn a multitude of timing tricks with our classic guide

As a musician of any experience and ability level, it surely won't have escaped your notice that the concept of time is intrinsic to all forms of music, as language is to literature and colour is to painting. However, ask yourself how much you really think about it, in a truly contemplative sense, as opposed to just figuring out where the taps of that eighth-note delayed lead synth are going to land in relation to your drum track.

With the vast majority of modern Western music being composed in 4/4 and adhering to a fairly standard set of unwritten rhythmic 'rules' in order to maximise its mass appeal, anything that breaks from the norm and finds its own groove can surely only be a good thing, and when it comes to bending musical time, there are many ways in which this simple goal can be achieved.

With all of this at the forefront of our mind, we invite you on a journey into the furthest reaches of the fourth dimension to explore a galaxy of techniques for bringing a touch (or an enthusiastic punch!) of temporal intrigue, spectacle and wonkiness to your tunes. We'll cover everything from gravity-defying odd time signatures and brain-melting polyrhythms to dancefloor-igniting tempo changes and groove-shifting beat displacement, all with the aim of sparking your creativity.

We promise to keep things as 'unacademic' as possible – there'll be no heavy theory or stave-based notation to negotiate, just practical DAW-based walkthroughs and tips, complete with audio examples, all designed to get you thinking more deeply about rhythm and metre, and producing more interesting tracks as a result, no matter what genre you're working in. And remember, these timing tricks aren't just for drum beats – they apply equally to any musical part, from basslines and synth riffs to vocals and lead guitars!

Now, time waits for no man, and the clock is already ticking, so let's get to it!

Tempo and time signature

Central to the concept of musical time are tempo and time signature. The first of these couldn't be more straightforward: tempo is simply speed, expressed as a number of beats per minute (BPM). While it might be easy to understand, though, the tempo of a track plays a huge part in defining its feel and character, and even placing it in a particular genre or sub-genre: dubstep at 140bpm vs drum & bass at 170bpm vs 2-step at 130bpm, for example. We'll come back to tempo shortly, but time signature warrants more of an explanation...

About time

If you've been producing music for a while and have never heard the term 'time signature' before, you're probably making all your tracks in the DAW default of 4/4. There's absolutely nothing wrong with this – indeed, the vast, vast majority of Western music today is in that stalwart sig – but by eschewing 'odd' alternatives such as 3/4, 5/4 and 6/8, you're missing out on not only a wealth of powerful compositional possibilities, but also a whole world of time-based fun.

So, what, exactly, do those two numbers actually describe? Although unofficially standardised in text as X/Y, in musical notation, the time signature (which sits in between the clef and the key signature) is written as X on top of Y, kinda like a fraction. Requiring only the most basic understanding of music theory to get a handle on, the meanings of X and Y really aren't as mystical or complicated as some may have led you to believe: the top numeral tells you the number of 'beats' in a bar, while the bottom numeral (the denominator) gives the note value of each of those beats. OK, we just made it sound more complicated than it actually is. Allow us to illustrate...

A bar of 13/8 could be thought of as three sets of three eighth-notes followed by a set of four

To take 4/4 as the most obvious example, the first 4 indicates that there are four beats in the bar, while the second (the denominator) tells us that the note value is a 1/4-note – also known as a quarter-note or crotchet. Thus, 4/4 means four 1/4-note beats to the bar. In exactly the same way, the time signature 9/8 sees each bar divided up into nine eighth-notes (aka 1/8-notes or quavers) for the purposes of performance, notation and – most pertinently for the computer musician – sequencing. Speaking of 9/8, time signatures that divide the beat into three (three groups of three eighth-notes each, in that particular case) rather than two are called 'compound' signatures, and they essentially define a track as having a triplet feel.

Simple maths

Ultimately, while most of the time you will indeed find yourself writing in 4/4, dropping a track into a so-called 'odd' time signature for a breakdown or middle eight, say, or even writing the whole thing in one, can pay rich creative dividends. An odd time signature is any one that has an odd number of beats in the bar – 5/4, 7/8, 11/16, etc. Yes, it would be true to say that 3/4 and 9/8 (and indeed, other time sigs with 3 and 9 'on top') strictly qualify as odd time signatures, but they don't sound 'odd' in the same way that 5, 7, 11 or 13 beats do.

Much of the time, composers will handle odd time signatures by simply subtracting one from the number of beats, dividing the remainder up into its obvious phrasing, then adding one back onto the last phrase in the bar. A bar of 13/8, then, could be thought of as three sets of three eighth-notes followed by a set of four, giving the feel of three sets of triplets followed by a non-triplet four-note phrase. This is by no means written in stone, though, and you could equally subdivide your 13/8 bar into two four-note phrases and a five-note one. Let's look at some examples of odd time signatures in action...

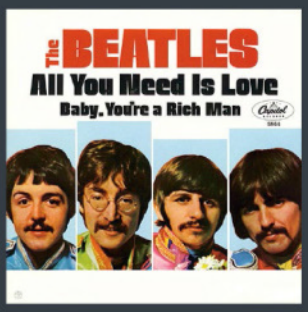
Recommended odd time listening

THE DAVE BRUBECK QUARTET
TAKE FIVE

When you think of famous tunes in odd time signatures, *Take Five* is probably the first one that comes to mind – and understandably so, since it still stands as the best-selling jazz single in history. It's got an awesome 5/4 groove, an unforgettable piano riff, a cracking drum solo and a beautifully considered approach to its time signature. What's more, the album from which it's taken, Dave Brubeck's seminal *Time Out*, packs in six additional non-4/4 tunes. Essential listening.

DIONNE WARWICK/ ARETHA FRANKLIN
I SAY A LITTLE PRAYER

While the chorus of this Burt Bacharach number is definitely in odd time – 11/4, phrased 4/4, 3/4, 4/4 – it's the less obviously wonky verses that make for the weirdest sections. Comprising two bars of 4/4, followed by a bar of 10/4, then two more bars of 4/4, on paper they look like a temporal train wreck waiting to happen, but in practice they work nothing short of perfectly. Oh, and the bridge between chorus and verse is a quirky three bars of 4/4, although that feels positively conventional in comparison.



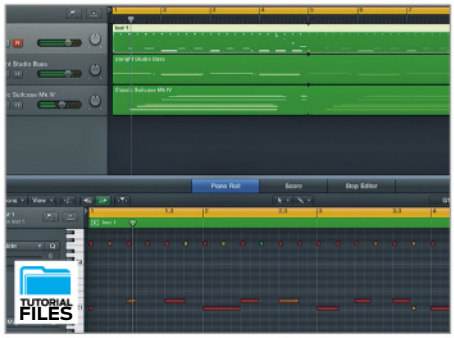
THE BEATLES
ALL YOU NEED IS LOVE

1967 was the year of mixed up time signatures in pop, and The Beatles' take on the concept turns out to be even more out-there than Bacharach's. Everyone knows this anthemic classic, but what you may not have noticed immediately is its off-the-grid nature, laying out two bars of 7/4, two bars of 4/4 (or one bar of 8/4, depending on who you ask) and another bar of 7/4 for the verses, then seven bars of 4/4 capped with one bar of 6/4 for the chorus. Utterly eccentric, totally awesome.

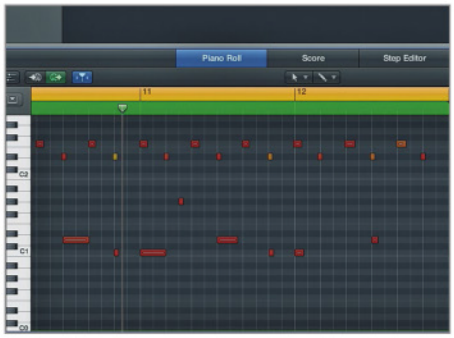
PINK FLOYD
MONEY

Although in technical terms not in the same over-indulgent, pyrotechnic league as the time-bending shenanigans of prog rock peers King Crimson and Genesis, Floyd's less 'muso' approach to odd time is arguably more successful. Money is in 7/4 (and 4/4), and like *Take Five*, it's easy for even the most cloth-eared listener to keep up with, thanks to the cash-clinking tape loop intro that sets out its stall, and the aurally intuitive phrasing used throughout.

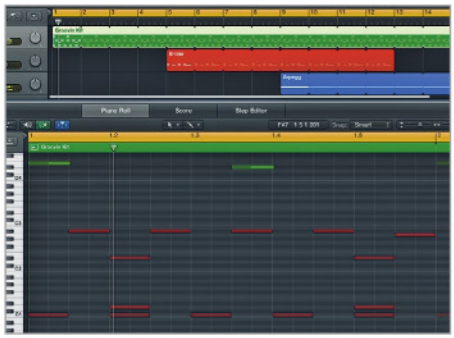
> Step by step 1. Odd time signatures in action



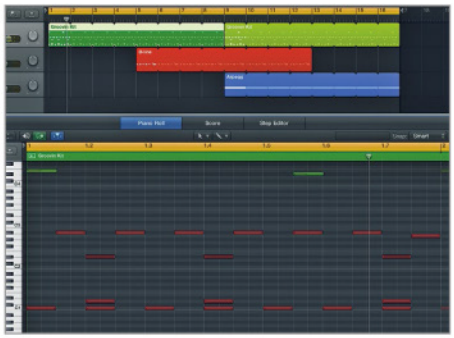
1 > Although we don't really consider 3/4 to be an odd time signature due to its conventional feel, in literal terms it is one, so here it is: 3-4 time.wav (our audio clips are available to download with this issue). With its characteristic lilt and swing, it can seem anachronistic these days unless handled with care, since the main thing it brings to mind is that decidedly old-school urban dancefloor movement, the waltz.



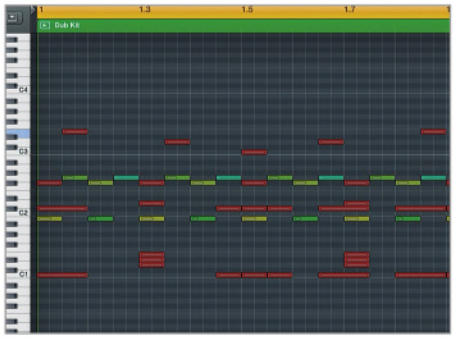
2 > Although you might think it would sound the same as 3/4, 6/8 divides the bar into two sets of triplet eighth-notes, rather than one set of quarter-notes, which implies a different sort of rhythm. Demonstrated in the second tutorial audio clip, 6-8 time.wav, it could be a double-time, march-like feel, or a half-time beat.



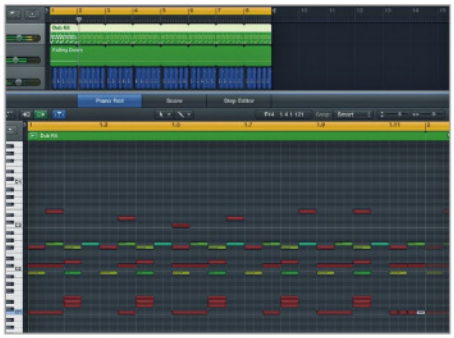
3 > The 5/4 time signature adds an extra beat to your regular bar of 4/4, generating a staggered, 'drunken' sort of rhythm. In the 5-4 time.wav clip, we've taken a standard four-to-the-floor groove and made it five-to-the-floor, with the second snare hit shifted over a beat to the right.



4 > Now, after converting our 5/4 groove to a 7/4 one, called 7-4 time.wav, our beat comes across as a four-beat phrase followed by a three-beat one. Since our bar is now quite lengthy, we can effortlessly drop to a half-time feel by thinning out and moving the snare hits.



5 > The longer the bar, the more space you have within it to mix implied time signatures. Our 11/4 example here (11-4 time 1.wav) sounds like two bars of 4/4 and one of 3/4, but we could have gone for three bars of 3/4 and one of 2/4, say.



6 > Finally, 11-4 time 2.wav is an alternative 11/4 phrase, comprising a DnB/2-step pattern for the first ten beats of the bar, then a one-beat kick drum fill on beat 11. Odd time signatures don't have to be overtly bonkers – they can also be used subtly, adding a twist to the end of an otherwise standard pattern.

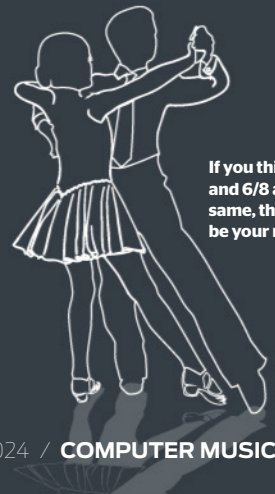
Anomalies and strangeness

You might well be wondering at this point what the practical difference between certain time signatures is. Why, for example, state a time signature of 6/8 rather than 3/4, when both have the same number of quarter-note beats (three) in the bar? Like many things in music theory and notation, there is no scientific answer (this is an art form, remember!) – rather, each brings with it its own style of notation and suggests a particular feel.

To resolve that specific example, 3/4 would be used for a waltz, while the pacier-by-implication (eighth-notes being twice as 'fast' as quarter-notes) 6/8, with its two triplets to the bar, would be more applicable to a march. Your DAW's quantise grid will be of more than high enough resolution to render this difference irrelevant, except for the speed of the metronome, which will click away at the chosen note value and obviously encourage a different vibe when recording, depending on whether it's sounding three quarter-notes or six eighth-notes in the bar. If you're not sure if your piece is in 3/4 or 6/8, consider how you would tap your foot or fingers along to the beat – the answer should immediately become apparent.

In reality, the only signatures you're ever likely to encounter for which the above is really relevant are in fact 3/4 and 6/8, thanks to the aforementioned waltz/march variation of the tutorial's first two audio examples, which is significant. While it might well seem like a good plan to put your latest retro jungle track together in 8/8 because of its double-time sensibility, we'd be very surprised if you actually did, since it really is exactly the same thing as using a load of eighth-notes in 4/4.

Incidentally, although non-power-of-2 denominators (referred to as 'irrational' time signatures) such as 4/6, 8/10 or 5/12, say, are theoretically possible, they're extremely unusual, largely academic and unsupported by any DAW we know of, so you can pretty much forget about them.



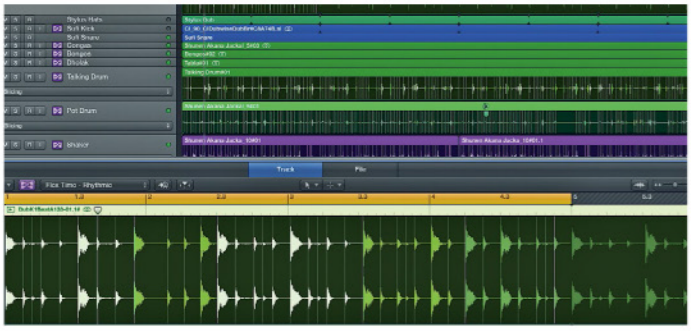
If you think 3/4 and 6/8 are the same, this could be your next gig

Manipulating time signature and tempo

There are many reasons why you might want to work tempo and/or time signature changes into a track. The most obvious one would be to dramatically change the feel of a song; switching from 4/4 to any odd time signature is about as dramatic a temporal change as you could ever hope to pull off mid-track. And the signature to which you change doesn't have to be an odd one, either. Throwing a 'fill' bar of 2/4 into a 4/4 project can be a more subtle way of throwing the listener a curveball.

Tempo changes are considerably more varied in their potential usage scenarios. Of course, like time signature changes, they can be just the thing to push a track in a totally new direction, whether applied suddenly (blasting into a half-/double-time section, perhaps) or gradually changed over a number of bars.

Some DAWs feature the ability to slave the project tempo to that of a chosen audio clip, enabling, for example, a live drum loop that varies in tempo to be set as the master for every other track in the project to follow (assuming the DAW in question also facilitates automatic timestretching of the audio on those tracks). Gently raising and lowering the tempo as a track progresses (known as 'rubato') in order to add pace to choruses and ease off in the verses is an age-old technique dating back centuries – originally the job of the orchestral conductor, these energy-giving fluctuations can now be drawn right onto the tempo track in your DAW.

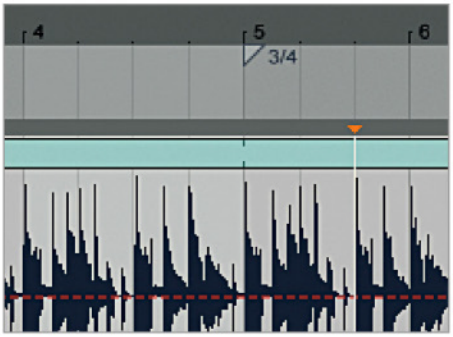


Timing tricks are straightforward to employ in most DAW-based productions

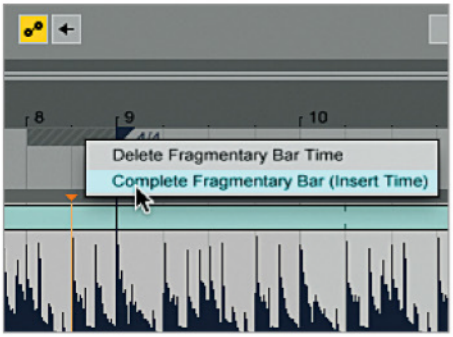
In the next two walkthroughs, we'll show you how to apply both time signature and tempo changes in Ableton Live. If you're using another DAW, the creative principles will be the same, though you may need to consult its manual to find out exactly how it's achieved.

> Step by step 2. Time signature changes in Ableton Live

1 > Ableton Live can work in any time signature with a denominator of 1, 2, 4, 8 or 16 and a numerator of 1-99. Yes, 99! The time signature is entered into the Time Signature field next to the metronome, and automating changes to it is a piece of cake. Let's look at how – start a new project in Live, which will default to 4/4.



2 > Right-click anywhere in the Scrub area below the timeline and select Insert Time Signature Change. A marker is created with an active field into which you can enter a time signature as two numbers separated by a slash, space, comma or full stop. When that point in the project is reached, it'll switch to that time signature – we've gone for 3/4.



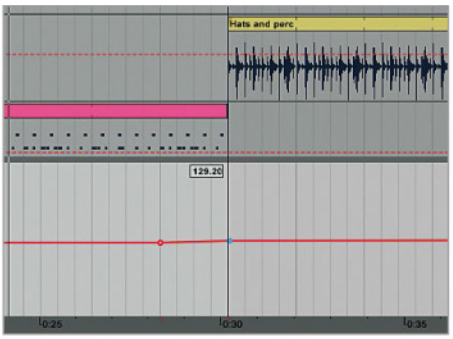
3 > If you add your change *within* the preceding bar, as opposed to on the bar line at the end of it, a 'Fragmentary Bar' will be created to fill the space, indicated by a dark grey strip. This can be left as is, or you can right-click it and select Complete Fragmentary Bar (Insert Time) to extend it back up to a full bar of the previous time signature.

4 > Similarly, if you edit a time signature marker (right-click and select Edit Time Signature) that is followed by another marker, such that the first section no longer fits exactly in the same timespan, a fragmentary bar will be placed at the end to fill the 'orphan' space. Keep it there or fill it out – the choice is yours.

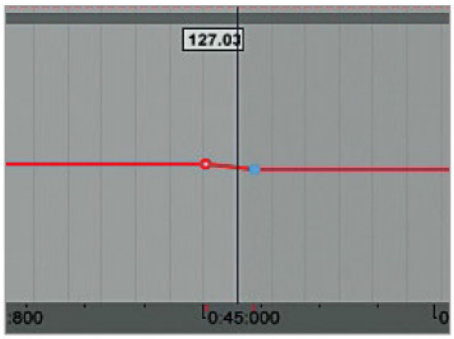
5 > Be aware that changing time signature has no bearing at all on tempo or the warping of audio clips, which will simply roll on regardless, as originally recorded/inserted. So, if you've already laid out your 4/4 drums, for example, you'd need to edit them to fit the new 3/4 'timescape', as we've done in 'Edited drums.wav'.

6 > Rather cleverly, Live even enables time signature changes in the Session view. Simply rename a Scene to the time signature you want to have it automatically switch to when that Scene is triggered. Note that any Scene not named in this way will play back at the last signature chosen.

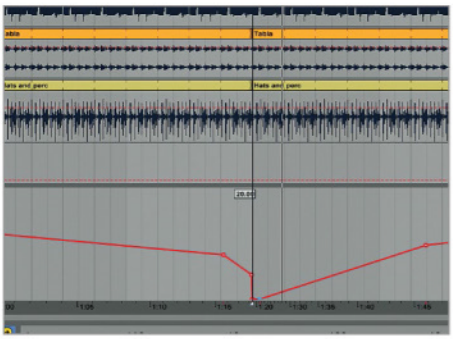
> Step by step 3. Tempo changes in Ableton Live



1 > Our 'No changes.wav' clip comprises verse, bridge, chorus and verse parts, eight bars each at a constant 127bpm. In the Up into chorus.wav, we've taken the bridge and chorus only, raising the tempo from 127 to 129 during the transitional drum break, then increasing it to 130bpm as the chorus progresses. This increase drives the song along, though we wouldn't recommend more than a 3bpm change.



2 > While the one-bar drum break at the end of the bridge enabled us to 'hide' our tempo change over the course of the bar, we have no such camouflage dropping from the chorus into the second verse. It doesn't really matter, though, as the sudden drop from 130bpm back to 127bpm is perfectly effective, applying the brakes as we go back into the verse. (In the Down into verse.wav clip.)

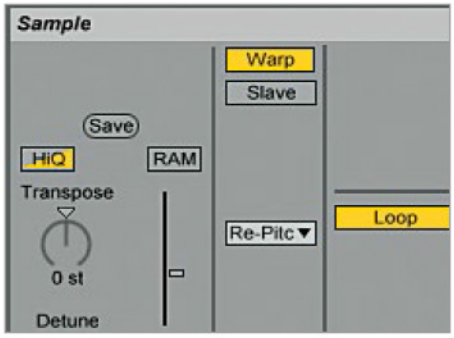


3 > Of course, there's nothing stopping you from ramping the tempo up and down more dramatically. In Long slowdown.wav, we're gradually slowing all the way down to 20bpm through the chorus section, then equally gradually coming back up to the full 127bpm to lead back into the bridge. It's retro, certainly, but done well, this sort of thing can have a great impact on the dancefloor.

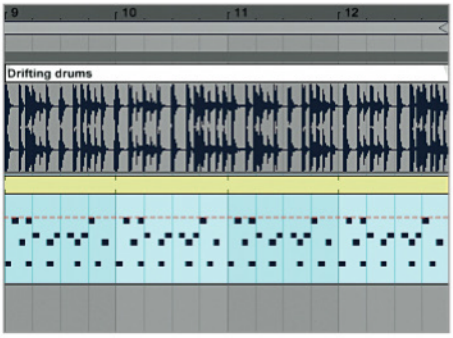
POWER TIP

>Flexi-time

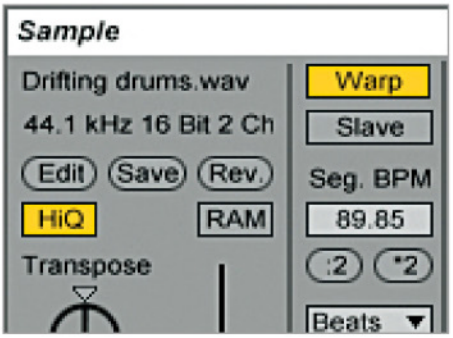
Live's project tempo parameter can be assigned to a knob or fader on your hardware MIDI controller for hands-on manipulation of BPM. It also makes a highly entertaining target for Max For Live's awesome LFO device, and ranging up to a maximum tempo of 999bpm, this particular combination is a great one for glitch and electronica producers. In fact, why not combine the best of both worlds, assigning LFO to tempo, and your MIDI controller to the LFO's Rate and Depth controls?



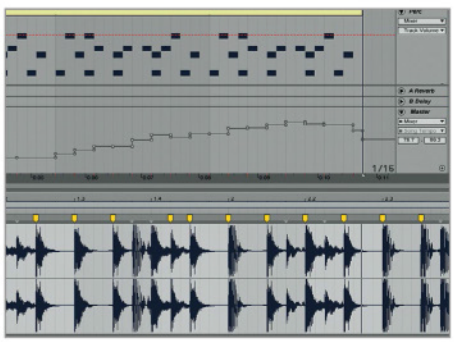
4 > So far, all of our audio clips in Live have been set to Beats (for the drums and percussion) and Tones (for the bass) warp modes. By changing everything to Pitch mode for our slow-down/speed-up section, we can get a pretty awesome turntable manipulation-style effect. We wouldn't generally stretch this out over such a long time, but to demonstrate, the 'Pitch mode.wav' clip makes the point.



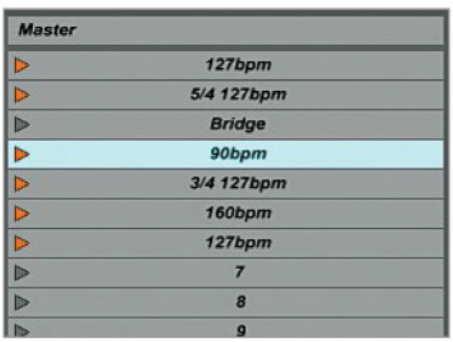
5 > Live can read the tempo of an audio clip and use it to create a tempo automation envelope so your project follows its timing. Here's a drum loop that slows down and speeds up throughout its four bars, and a perfectly in-time programmed percussion part that we want to match up to it. '5. Wonky drums.wav' showcases one, then the other, then both together.



6 > The first thing to do is warp the drum loop so that it lines up perfectly with the beat. With that done, click the Slave button in the loop's Clip View to make it the tempo Master.



7 > Reveal the Song Tempo automation lane on Live's Master Track and you'll see that a series of project tempo changes have been created to match the loop's tempo. Our percussion is now perfectly in time with our (totally out of time) drums. Right-click the Song Tempo envelope and select Unslave Tempo Automation to make it editable; but bear in mind that this puts the clip back into Slave mode.



8 > In the previous walkthrough, we showed you how to apply time signature changes in Live's Session View. It should come as no surprise that tempo changes can be triggered in the same way: simply name each Scene as the tempo you want it to switch to. Want to change the tempo and time signature at the same time? Of course you do! Just name your Scene using both – '5/4 127bpm', say.

Syncopation

Fundamental to just about all modern music (and a great deal of older material, too!), syncopation is – in literal terms – the placing of accented notes on the 'weak' beats of the bar. This is an extremely open-ended definition, however, given that those 'weak' beats could be considered to be beats 2 and 4 (which, technically, makes the common or garden backbeat a form of syncopation) or the eighth-notes that fall in between the four main beats of a bar of 4/4, or even the 16th-notes that fall in between those, and so on.

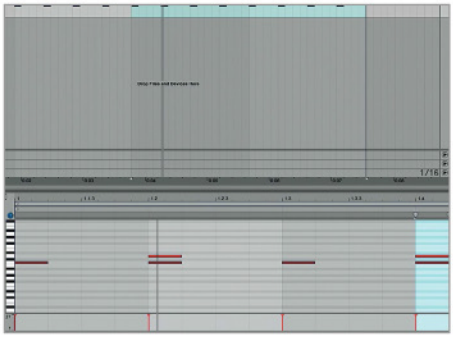
To put it in non-technical terms, syncopation is what puts the funk, groove, skank or other rhythmic 'feel' into a piece of music. Although it's been a quantifiable concept in all genres and styles since man first took stick to log, it's always been particularly important in dance music, which just wouldn't work without it – imagine, if you will, a dancefloor full of people just stomping along to beats 1 and 3. Not our idea of a fun night out. By dance music, we mean all the way back to the tribal beats of pre-history, then up through medieval music, classical, jazz, rock 'n' roll, and on to house, hip-hop, jungle, trance, electro, dubstep and all the rest of 'em.

We've no doubt that you're already using syncopation in your productions, whether you're conscious of it or not – you'd certainly know if you weren't using it, as your tracks would sound mechanical, entirely 'on the beat' and rhythmically dull. To make things crystal clear, however, we're going to walk you through some specific examples of it that will hopefully inspire you to think more about where you're placing those accents and, consequently, up the funk factor of your tracks.

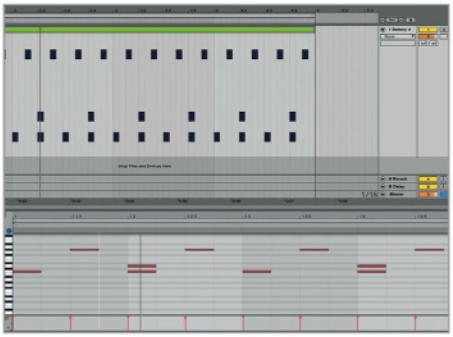
© Getty Images

Funk is built upon syncopation; James Brown would've been a decidedly stiff and ungroovy fellow without it

> Step by step 4. Basic syncopation of a drum part

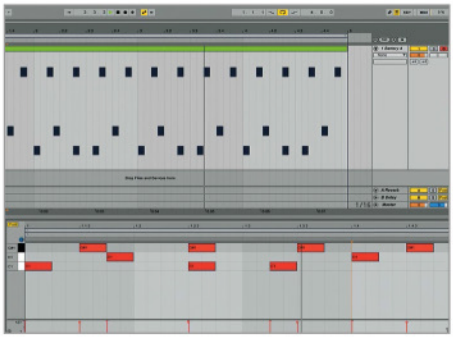


1 > Here's a truly simple example of syncopation: the good old backbeat! Our Backbeat.wav starts with a non-syncopated 4/4 kick drum pattern; then we syncopate it by adding a snare on beats 2 and 4. With beats 1 and 3 (the onbeats) considered the 'strong' beats in the bar, emphasising 2 and 4 (the offbeats) qualifies as syncopation, at least in a purely technical sense. It ain't that funky, though...

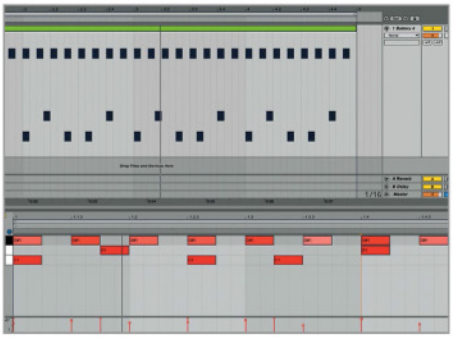


2 > We can syncopate our beat even further by adding an offbeat hi-hat. The term 'offbeat' has two meanings in music: beats 2 and 4 (in a bar of 4/4) and any note that falls in between the main beats. We go for the latter, placing a hat on every second eighth-note. Our drums are now syncopated on two levels. (Check out With hats.wav.)

3 > So far, so predictable. Let's turn our mundane 4/4 beat into something a little more interesting. There are countless ways in which we can do this, but one of the easiest is to just move one of the snare hits off the beat – we're really getting syncopated in '3. Move the snare.wav'.



4 > Every element of our drum kit can be syncopated, so let's move on to the kick drum. By syncopating the kick, as heard in our 'Move the kick.wav', we greatly up the perceived pace of our groove and give it a much more funky feel than we were getting with that rigid four-to-the-floor pattern.



5 > Finally, the hi-hats, which need to drive the groove along but can still be made more rhythmically interesting via the magic of syncopation. First we double up the hits, then we accent certain ones and quieten others, paying attention to their interplay with the kick and snare. Check it out in More hats.wav.

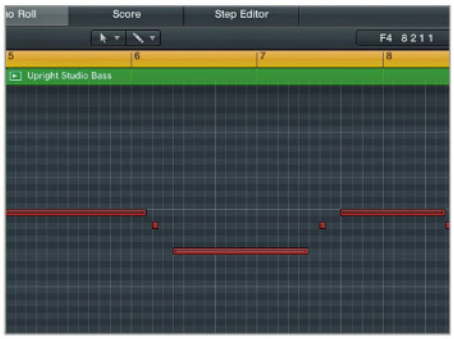
POWER TIP

>Solid foundations

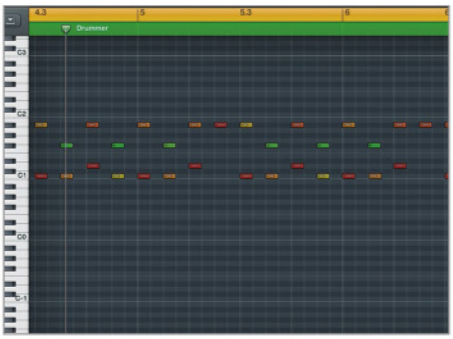
When working on syncopating drum patterns, don't forget that the first priority for drums most of the time is to absolutely nail the rhythm of your track to the floor, as it were. Be careful, therefore, not to detract from or overly 'lighten' the kick and snare with your offbeat flouncery; ghost notes are meant to lead into and out of your main snare hits, rather than replace them, for example. Of course, if overly busy is the vibe you're going for, feel free to disregard this advice!

> Step by step 5. Exploring deeper syncopation ideas

1 > Although syncopation is a rhythmic thing, melody and harmony can also come into the equation, thanks to our expectations of where 'strong' chords ought to fall within a bar. The 'Harmonic syncopation.wav' clip showcases a Rhodes part in which the two chords are syncopated at the end of each four-bar phrase to strange psychological effect.



2 > Extending or shifting notes or chords within a phrase so that they cross over into the next bar is another form of syncopation that can really change the feel of a track. '2. Crossing the bar.wav', has both the Rhodes part and the bassline crossing the bar lines at various points.

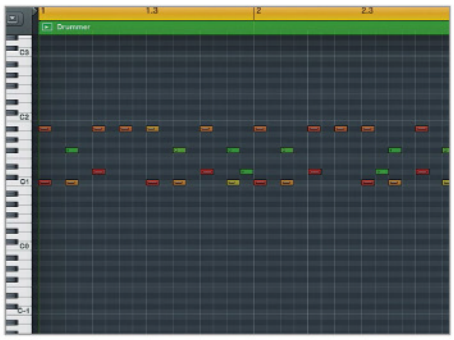


3 > Syncopating drum parts by moving or adding kick and/or snare hits and accenting the hi-hat is all well and good, but for maximum groove, you need to be adding ghost notes, too. These are quiet notes played by a drummer in between the main hits – usually on the snare – to add movement. Compare our 'Without ghost notes.wav' file...

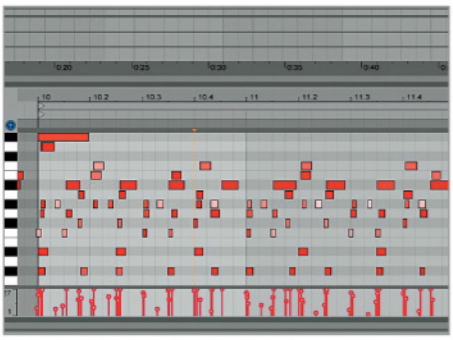
POWER TIP

>Everybody salsa

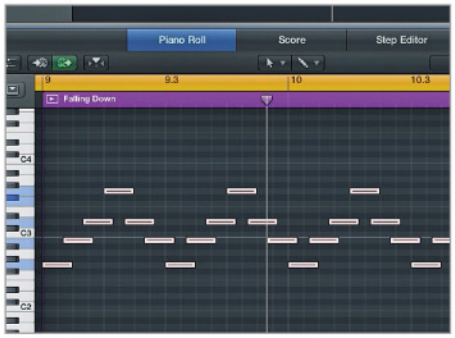
Many styles of dance music can benefit from a bit of sampled (or live!) Latin percussion. Even if it only comprises a few hits, a tastefully programmed conga, shaker, timbale or cowbell line can exponentially funk up any drum part. And don't worry about authenticity; while we'd certainly recommend getting your ears round some live percussion performances on YouTube to make sure your MIDI editing heads in the right direction, as long as you capture that essential syncopation and flow, you're golden.



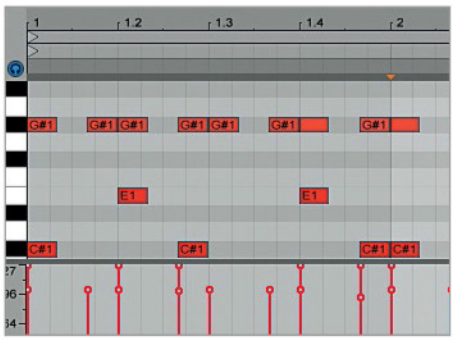
4 > ...to 'With ghost notes.wav'. This time, with ghost notes in place, (the green notes on the snare in Logic's Matrix editor – note D1). When it comes to feel and 'humanity', the difference between an acoustic drum track with ghost notes, and one without, can be like night and day.



5 > Perhaps the most intrinsically syncopated collection of genres of all, Afro-Cuban and Latin music weaves intricately layered percussion parts – each one heavily syncopated – together to create complex rhythms over which all the other equally syncopated instruments do their thing. 'Latin percussion.wav' is a Latin-esque percussion jam built up layer by layer.



6 > Although not traditionally considered syncopation in itself, the triplet – that is, three regularly spaced notes squeezed into the same time span as two (ie three eighth-notes played over the course of one quarter-note) – certainly spices up any non-triplet-based part. In the second half of the Triplet synth.wav, the synth arpeggio switches to a triplet pattern.



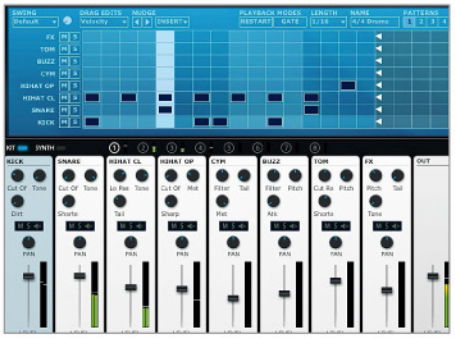
7 > By knocking out certain notes from a triplet pattern, a shuffling, swung feel can be created. In the first half of Triplets drums.wav, you can hear a straight triplet-based drum track. In the second half, every second note of three has been removed from the hi-hat part and snare fill – quite a difference!

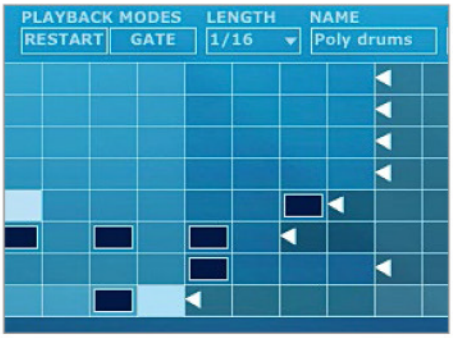
POWER TIP

>Triplet trouble

Unless you're composing in a triplet-feel time signature such as 6/8, exercise restraint with your triplets. As our synth audio example demonstrates, constant triplets against a regular non-triplet backing can easily become hard to listen to and somewhat bewildering for the listener. If working triplets into a part separates it from the overall groove of the track too much or for too long, shorten the triplet section or switch your MIDI editor back to its regular two/four-based grid.

> Step by step 6. Polymetres in FXpansion's classic Tremor





1 > Polymetre is the use of more than one time signature at the same time – usually on a per-instrument or per-element basis – but with everything playing at the same tempo. FXpansion's classic Tremor drum machine demos an ingeniously simple way of implementing polyrhythms. '4-4 drums.wav' is a simple one-bar drum pattern in 4/4.

2 > By dragging the white triangular pointers on each sequencer lane to the left or right, we can adjust the length of that lane's loop, effectively changing its time signature. In Polymetric drum kit.wav, the closed hi-hats are in 7/8, the open hi-hat is in 15/16, the kick drum is in 3/4, while only the snare remains in 4/4. Head-spinning stuff!

3 > All four lanes won't land on the 4/4 downbeat together again until bar 106 (without that 15/8 hi-hat part, mind, they'd sync up again every 22 bars). Finally, let's hear how it sounds with some added percussion in 8/4, 7/4, 5/4 and 11/8 – astonishingly, these eight elements won't all land on '1' together again until bar 2311! Check out 'Polymetric madness.wav'.

What are polyrhythms?

Without going into too technical waters, a polyrhythm is made from two or more rhythms that haven't been designed to work together, played at the same time.

We're classifying polymetre as a form of polyrhythm for the purposes of this tutorial, although some would argue that it doesn't really qualify, since a polymetric piece of music doesn't necessarily have to be polyrhythmic. Famous examples of polymetric music include Led Zeppelin's *Kashmir* (in which the drums and vocals are in 4/4, while everything else is in 3/4 for much of the track) and numerous tracks by Swedish metal outfit Meshuggah.

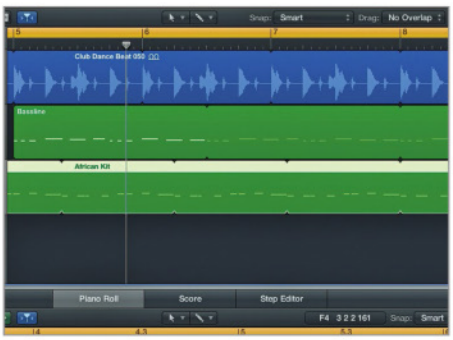
African and Latin music is heavily polyrhythmic in the more traditional sense of the word, as demonstrated by the Latin percussion section discussed on the previous page, in which each individual instrument basically 'does its own thing', emphasising and accenting specific beats and sub-beats (syncopation) in the same time signature, with the ensemble coming together to create its rhythmically intricate overall effect.

The influence of African and Latin styles on jazz, pop, rock and dance music is evident in the involvement of polyrhythmic techniques in all of them. No matter how simple or complex its execution, several disparate elements combining to form a single, multistrand whole is central to polyrhythmic composition and performance.

> Step by step 7. Polymetric MIDI and audio in a DAW

1 > Programming polymetric patterns in your DAW, and across multiple audio and MIDI tracks, can require some effort. Let's start with a conventional 4/4 drum part at 100bpm. Hear it in the '4-4 drum loop.wav' audio clip.

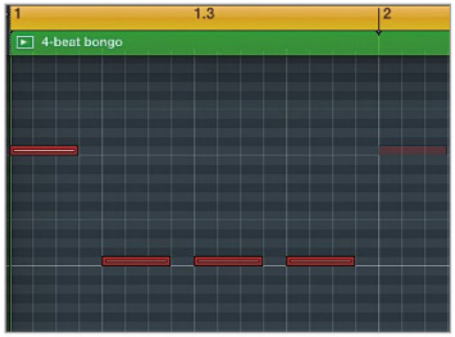
2 > Let's record a 3/4 bassline. First, switch the time signature to 3/4 and activate the metronome. Then mute the drum track so that it can't distract you back into 4/4, hit record and play your 3/4 bassline. '3-4 bass.wav' showcases our recorded two-bar pattern.



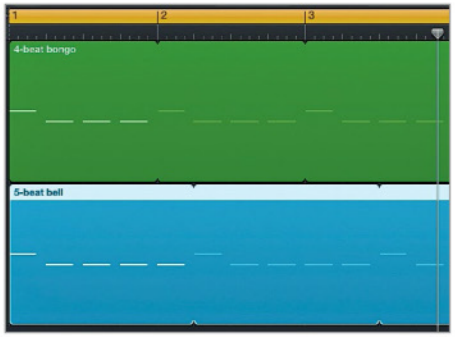
3 > We set our time signature back to 4/4 and record a bit of percussion, muting the bassline so as, again, to avoid getting dragged into a 3/4 feel ('4-4 perc.wav'). We then drag the end of the clip back to shave off the last eighth-note, putting the part in 7/8 time. Check it out in '7-8 perc.wav'.

4 > Finally, let's extend a 4/4 audio clip to make it 5/4. Import a 4/4 sample (we used '4-4 synth sample.wav'), set the time signature to 5/4 for visual clarity, cut your chosen 'extra' beat out of the sample, place it at the end of the clip and join the two together to make a five-beat phrase. Hear our final piece, with all elements together in 'Final polymetre.wav'.

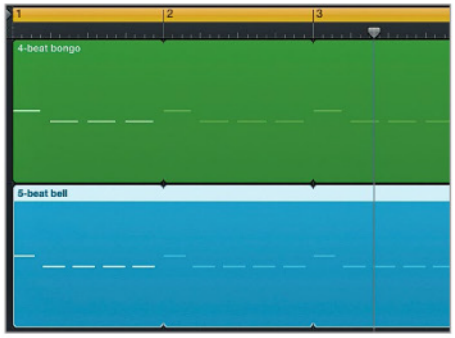
> Step by step 8. Programming cross-rhythms



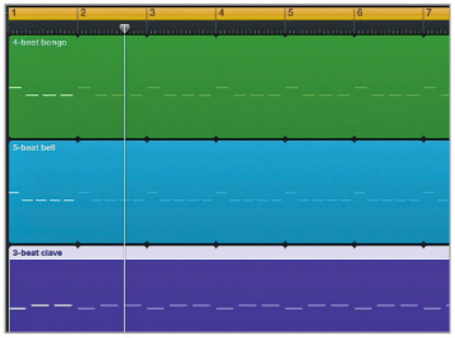
1 > Apart from doing it entirely manually (don't go there – it's a nightmare), the easiest way to work up cross-rhythms (read the sidebar opposite for more info) in MIDI is by timestretching clips. Let's make a 4-against-5 pattern. We start by programming a bongo hit on each beat of a 4/4 bar as in 'Audio: 4 beats.wav'.



2 > Next, we create a new track triggering a bell sample and draw a series of five quarter-note beats. Now we have a polymetric pattern: bongo in 4/4, bell in 5/4. This is not a cross-rhythm! To turn this into one, we need to somehow squash the bell part so that five equally spaced notes occur in the space of one bar.



3 > All that remains, then, is to timestretch the 5/4 bar down to the length of the 4/4 bar. How you do this will depend on your DAW, but in Logic Pro, it's a simple matter of holding down Alt and dragging the bottom right-hand corner of the bell clip back until it snaps to the barline. The file '3. 5 against 4.wav' demonstrates the two playing together.



4 > Finally, we'll add a three-beat clave pattern using the same technique. As before, we simply draw in a new clip comprising three quarter-note hits before timestretching it to fit the bar. Check out '5 against 4 against 3.wav'.

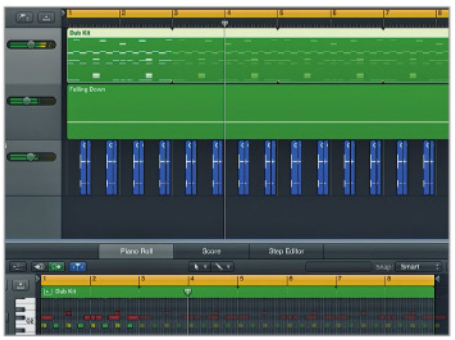
Crossing the streams

While a polymetre comprises multiple parts in different time signatures chugging along at the same tempo, and thus only coinciding on beat 1 (the downbeat) every certain number of bars, a cross-rhythm is what you get when two or more parts each comprise a different number of evenly spaced beats, all starting together on the downbeat of every bar.

If any one part was solo'ed, you'd hear a perfectly metronomic pattern at a tempo determined by the number of beats it contains, but play them all together and the results are anywhere from decidedly funky to seemingly chaotic. As the numbers of beats played by each part rises, things can get pretty complicated. While beating out 2-against-3 on the desk with your hands is a piece of cake, how about 13-against-7 or 5-against-21? Don't feel bad – even a seasoned drummer would turn pale at the thought of either of those two...

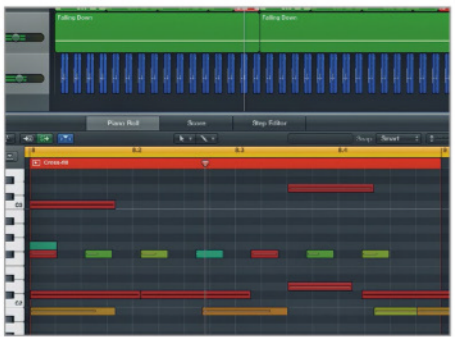
In the second walkthrough on this page, rather than just programming a regular pulse of≈evenly spaced notes for each instrument, we trimmed and timestretched the existing patterns from the main groove. Consequently, the *phrases* (and the tempo grids behind them) are cross-rhythmic, although the notes and accents within them are irregular and thus somewhat looser than your hardcore music theorist might be comfortable with. Hey, we're not trying to pass a music degree course here, you know!

> Step by step 9. Using cross-rhythms to create a fill



1 > It's all very well knowing how to make cross-rhythms, but what can you actually use them for? Well, you could base entire tracks on them, but that might get a bit... mind-altering. Perhaps a better application is to throw them into your regular 4/4 tracks as fills and punctuation. You might recognise 'Straight track.wav' from earlier.

2 > Let's turn the last bar of our eight-bar drum part into a cross-rhythm fill, with each element contributing an independent rhythm. If your DAW can't timestretch lines of notes within a MIDI clip, duplicate the last bar a few times and delete notes within the clips until each one only contains one kit piece line (tabla, hats, ride, kick, snare).



3 > We can now shorten or lengthen each kit element part to our required time-base, then use the previously described timestretching process to fit them all into the 4/4 bar, before merging them all back into one clip. 'Cross-rhythm fills.wav' contains our fill at original speed the first time round, then doubled in tempo the second time, which we think works better.

Beat displacement and mirroring

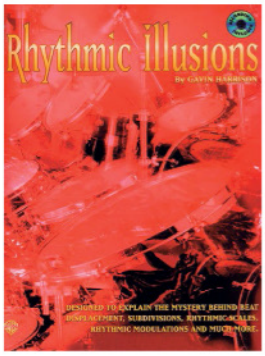
So far, we've aimed our rhythmic telescope at odd time signatures, tempo manipulation, syncopation and polyrhythms, but perhaps the easiest (and often most effective) of all temporal tricks would be in the realm of beat displacement. As the name suggests, this is the technique of shifting patterns forwards or backwards in time so as to knock them off the beat, the idea being to make the listener think that some dramatic change has been made to the rhythm, before resolving the pattern by putting it back how it was.

The amount of displacement applied will depend to some extent on the layout of the pattern (there's little point displacing a four-to-the-floor kick drum by a quarter-note, for example), but you'll usually want to keep your displacements to a 16th-, eighth- or quarter-note shift – smaller note values (or 'in between' ones, such as a 24th-note) will see you straying into experimental territory. Feel free to go there, though, if that's your bag...

Obviously, beat displacement is most effectively used on drum tracks, since the whole point is to shift the feel of the underlying groove of a track, predominantly defined by the ol' tubs. However, highly effective changes of pace and direction can be spoofed by knocking basslines and melodic parts off-axis, too – or how about pushing your lead line forwards at the same time as kicking the drums backwards, while leaving your lead line or vocal where they are?

And although not strictly speaking a displacement, mirroring the beat horizontally can have a similar effect. We'll look at some of these techniques in the following walkthroughs.

Incidentally, although the concept of beat displacement is presumably as old as music itself, the credit for formalising it goes to Brit drumming legend Gavin Harrison (Porcupine Tree, King Crimson, Lewis Taylor and many, many more), who explores the concept – amongst may others – in depth in his 1996 book *Rhythmic Illusions* (originally a tutorial series in *Rhythm* magazine). Although absolutely intended as a manual for drummers, MIDI programmers (who know how to read music!) looking to really take their percussion patterns to new and extremely esoteric places should seek this seminal text out; it's still available through the usual online book retailers.

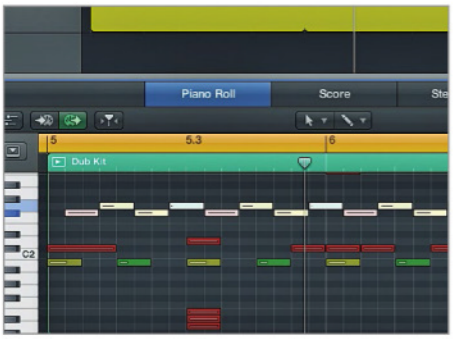


Pull the rhythmic wool over your listeners' ears with Gavin Harrison's *Rhythmic Illusions*

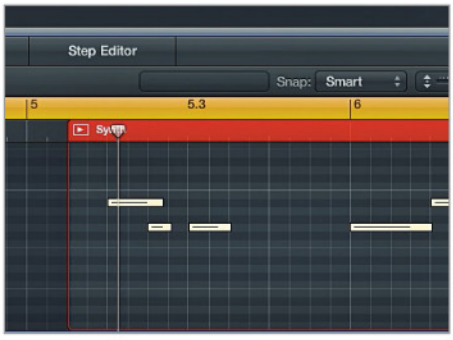
> Step by step 10. Beat displacement

1 > Here's a dubby groove with a synth line on top. Our first displacement is very straightforward: in the section from bar 5-9, we just drag the left hand edge of the clip to the right until it meets the first snare (two beats), then move the whole clip back to the left. The result ('2-beat shift.wav') is the snare and main kick drum beats swapping place, 'reversing' the groove.

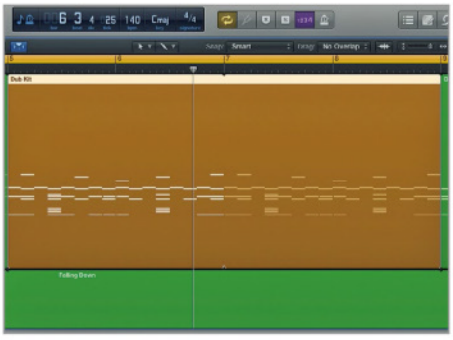
2 > We can get an even wonkier vibe by shifting the drums to the left by just an eighth note. This really pulls the beat away from the main groove, most notably with the hi-hat accent at the end of the phrase. ('Eighth-note short.wav').



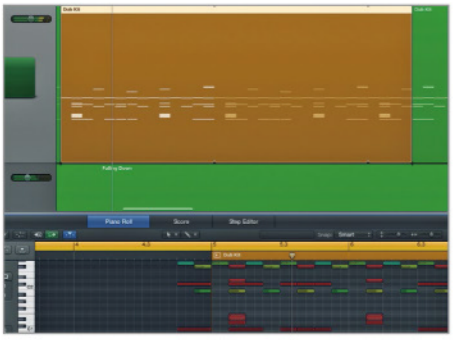
3 > 16th-note shifts need to be applied with care, as they can knock the whole rhythm off-centre, creating a 'sea-sick' feel. In 'Shifted hats.wav', only the hi-hats have been moved, and although it certainly breaks up the flow of the drums from bar 5 to bar 9, it doesn't make for particularly easy listening.



4 > As mentioned above – and, indeed, as with all the other rhythmic tricks we've explored throughout this article – beat displacement doesn't only work on drums. By nudging our synth line back an eighth-note, its relationship to the pulse changes and the track takes on a wholly different character. Hear it for yourself in 'Displaced synth.wav'.



5 > In step 1 we 'pseudo-reversed' the groove by shifting the snare over to beats 1 and 3, but by literally reversing the beat (using Logic's MIDI Transform » Reverse Position function), we get a rather different effect, and not – it must be said – quite what we were expecting before we hit the Operate button.



6 > Finally, we bring the whole exercise to its logical conclusion by displacing our reversed beat an eighth-note to the left. As you now hopefully appreciate, making profound changes to a drum track really can be as easy as sliding it left or right by a musically meaningful amount. Check out 'Reversed and displaced.wav'.

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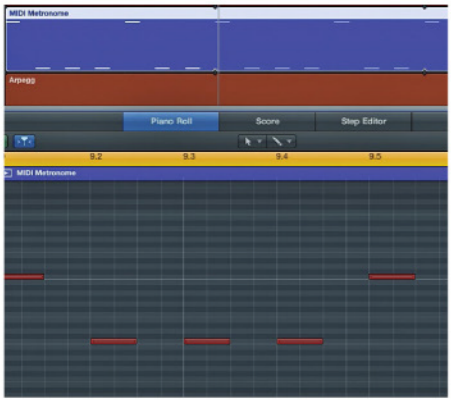
Rhythmic tips and tricks

01 There are several tried and tested ways of smoothly segueing into a time signature or tempo change, rather than just instantly switching to it, which can be jarring. One of them is to end the bar before the change with a fill (most likely on drums, but any attention-drawing instrument should be up to the job) that heavily features triplets. The reason this can make for an effective transition is that the triplets work as a disruptive, ear-catching intermediate stage of rhythmic oddness leading into the more obvious rhythmic structure that follows.

02 Another good way to lead into a tempo change is to again put a fill at the end of the phrase preceding the switch (not necessarily featuring triplets, unless you want it to, of course) and make the tempo change at the start of that bar, rather than the one at the start of the new section. This could potentially work with a time signature change as well, if your fill is designed cunningly enough to feel like a transition from one to the other. Or, if you're switching to a 'longer' time signature (from 4/4 to 5/4, say), try putting a fill at the end of the transitional bar that just makes up the extra time (one beat in our 4/4-to-5/4 example).

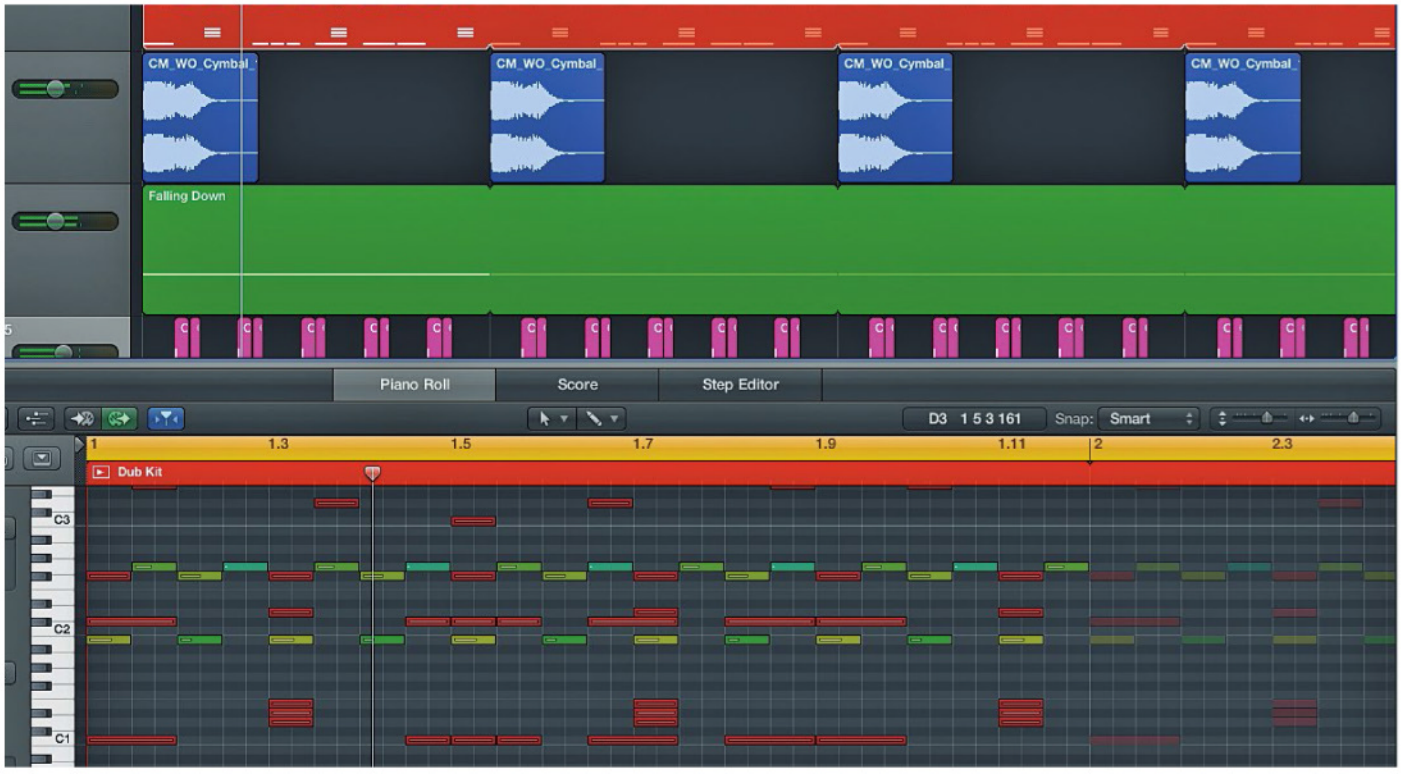
03 When working in odd time signatures with a numerator greater than 4, it can be very helpful to have the metronome emphasise the phrasing of the bar as well as the downbeat – beat 5 in 7/4, for example, or beats 4 and 7 in a bar of 9/8. It might be that your DAW's metronome doesn't allow for this, though, so why not program your own click track in MIDI, with different sounds, pitches or levels for these emphasised beats, using a sampler or drum machine as a sound source?

04 If you play an instrument but just can't get your head or fingers around playing the thing in odd time signatures, try recording your riffs and other parts in good old 4/4, then creatively cut and paste to edit them into your desired signature. For example, record two bars of a 4/4 guitar riff as you normally would, then chop off the last (or first) beat of the resulting audio clip to turn it into a 7/4 riff of one bar. Obviously, it's more satisfying and effective



If your DAW's metronome won't cut it, you can craft your own to help you keep in your newly jaunty time

05 If that odd time signature or polymetric section of your latest track is proving a bit too disorientating, making it hard for the listener to follow the groove, there's no shame in placing an obvious sonic marker of some sort on the downbeat (beat 1) of your pattern or riff – a crash cymbal or piano stab, perhaps.



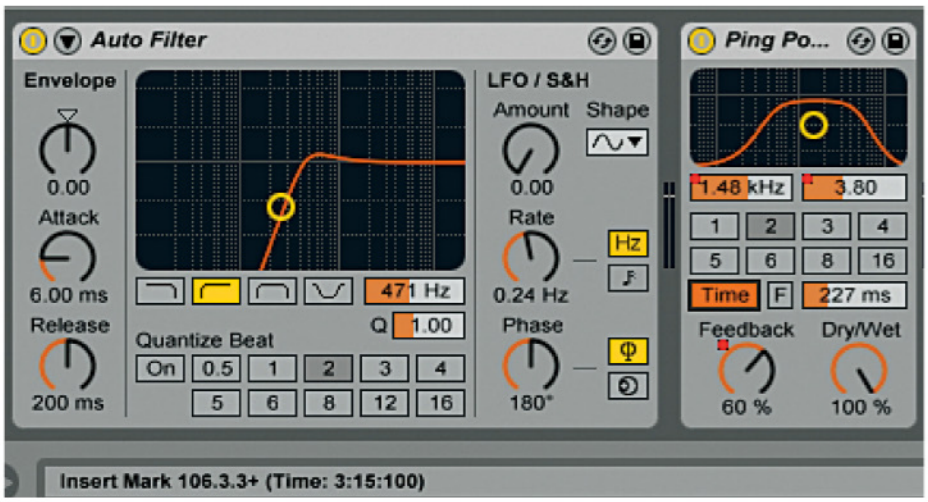
If your rhythmic offerings are a bit too complicated to be easily understood, emphasising the downbeat with a crash cymbal can keep listeners grounded

if you can actually play your odd-time parts for real, but until you get those chops up to scratch, editing in your DAW will do the job just fine.

06 We mentioned Swedish metal band Meshuggah earlier, but their particular approach to polymetre is worth highlighting again, since it works so well with 'beat-driven' music and has been imitated by so many other acts. Their particular approach often involves the drummer combining time signatures on various elements of the drum kit. For example, with a simple 4/4 rock beat rolling on the hi-hat and snare, a 3/4 or 5/4 repeating figure might be placed underneath it on the kick drum(s), with the downbeats accented on crash cymbals. Polyrhythms between individual instruments are one thing, but the drum kit (acoustic or electronic) is essentially a self-contained polyrhythm machine, so make it the first port of call in your mixed-meter adventures.

07 When recording new parts and punching in on existing ones, many musicians actually prefer a straight 4/4 count-in, even when playing in odd time signatures. Fortunately, DAWs often offer the option to tailor the count-in metronome behaviour, enabling you to keep the regular 4/4 count-in rather than a potentially confusing 9/8 one, say. So, if your drummer is having trouble feeling the downbeat of a 5/4 bar with a 5/4 count-in, try changing the latter to 4/4 and see if that helps. If your DAW doesn't offer that degree of metronome adjustment, you could always place a four-beat MIDI clip just before the punch-in point and route it to a sampler loaded with a metronome-type sound instead.

08 You can create the illusion of using weird time signatures by turning the snap-to-grid function off in your DAW and avoiding the grid to whatever extent works for the particular track in question. Even though this won't result in any actual odd time signatures (unless you're incredibly clever in your programming and bloody-minded enough to do it that way rather than actually switching to an odd time signature!), the effect on the listener will be similar – slightly disorientating and utterly groove-transformative. Classic '90s IDM is often held up as an example of the use of odd time signatures, but actually, a great deal of it is in plain old 4/4. Producers such as Squarepusher, Boards of Canada and Aphex Twin were just very good at programming MIDI in a way that makes the underlying sequencer grid extremely hard to perceive and follow, often weaving chaotic and intricate grid-shunning drum parts around more conventional synth parts.



Throwing off the rhythmic yoke of your DAW's host synchronisation leaves you free to experiment

09 While virtual instruments and effects automatically syncing LFOs, delay times and other parameters to the tempo of the host DAW is a distinct advantage over their real-world counterparts, it can actually be creatively liberating to turn it off. By running your time-based modulators and processors 'free' rather than slaved to the grid, all sorts of polyrhythmic possibilities reveal themselves – running multiple LFOs of a synth at different unsynced speeds, for example, or generating off-kilter, evolving rhythms from a single sound source by running it through an unsynced filter delay.

10 When programming gradual tempo shifts, remember that the transition does not have to be a linear one consisting of a straight line between tempo A and tempo B. Some DAWs have features to create curved transitions between automation nodes, but if not, you can add more nodes along the length of a straight line and adjust their positioning to approximate a curve. While there's nothing wrong with a linear transition, a curve can sometimes sound more natural or appropriate, so give it a go.

11 We've not delved into the use of groove, swing and shuffle in this tutorial; while timing is intrinsic to these, they arguably do not constitute rhythms in themselves but rather are temporal/dynamic modifications that can be applied (or not) to a given rhythm. That's not to belittle the importance of groove, though; indeed, we've delved right into it on numerous occasions. You can go right back to **cm**174's *Into The Groove* for one of the first, but there's also the more recent *Ultimate Beats* feature in **cm**313 and *Make Better Beats* in **cm**326. Combine such teachings with our *Radical Rhythms* and you'll be well on your way to metrical nirvana! **cm**

12 Since the vast majority of clips in your sample library will be in 4/4 time, it makes sense to label those that aren't with their time signature: "5/4 Funky Drums.wav", for example. Similarly, within projects featuring time signature changes, labeling MIDI clips by signature can help keep confusion to a minimum.

Name your unconventionally timed samples appropriately

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cm/reviews

The latest computer music gear tested and rated!

72 ABLETON LIVE 12

A major update from the world-beating DAW is always a bit like Christmas come early (/late). What new gifts await?

Our promise

We bring you honest, unbiased appraisals of the latest computer music products. Our experts apply the same stringent testing methods to all gear, no matter how much hype or expectation surrounds it.

What the ratings mean

1-4 A seriously flawed product that should be avoided

5 This product's problems outweigh its merits

6 A decent product that's only held back by a few flaws

7 Solid. Well worth considering

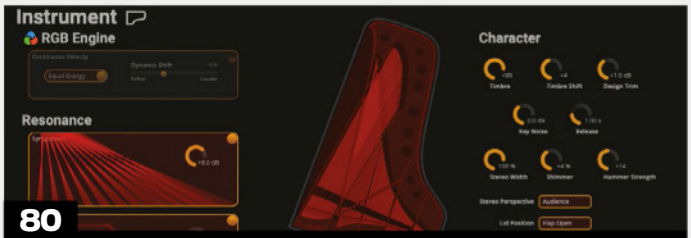
8 Very good. A well-conceived and executed product

9 Excellent. First-rate and among the best you can buy

10 Exceptional. It just doesn't get any better than this!

76 BLEASS Arpeggiator

78 Cherry Audio Pro Soloist



80 Synthogy Ivory 3 German D

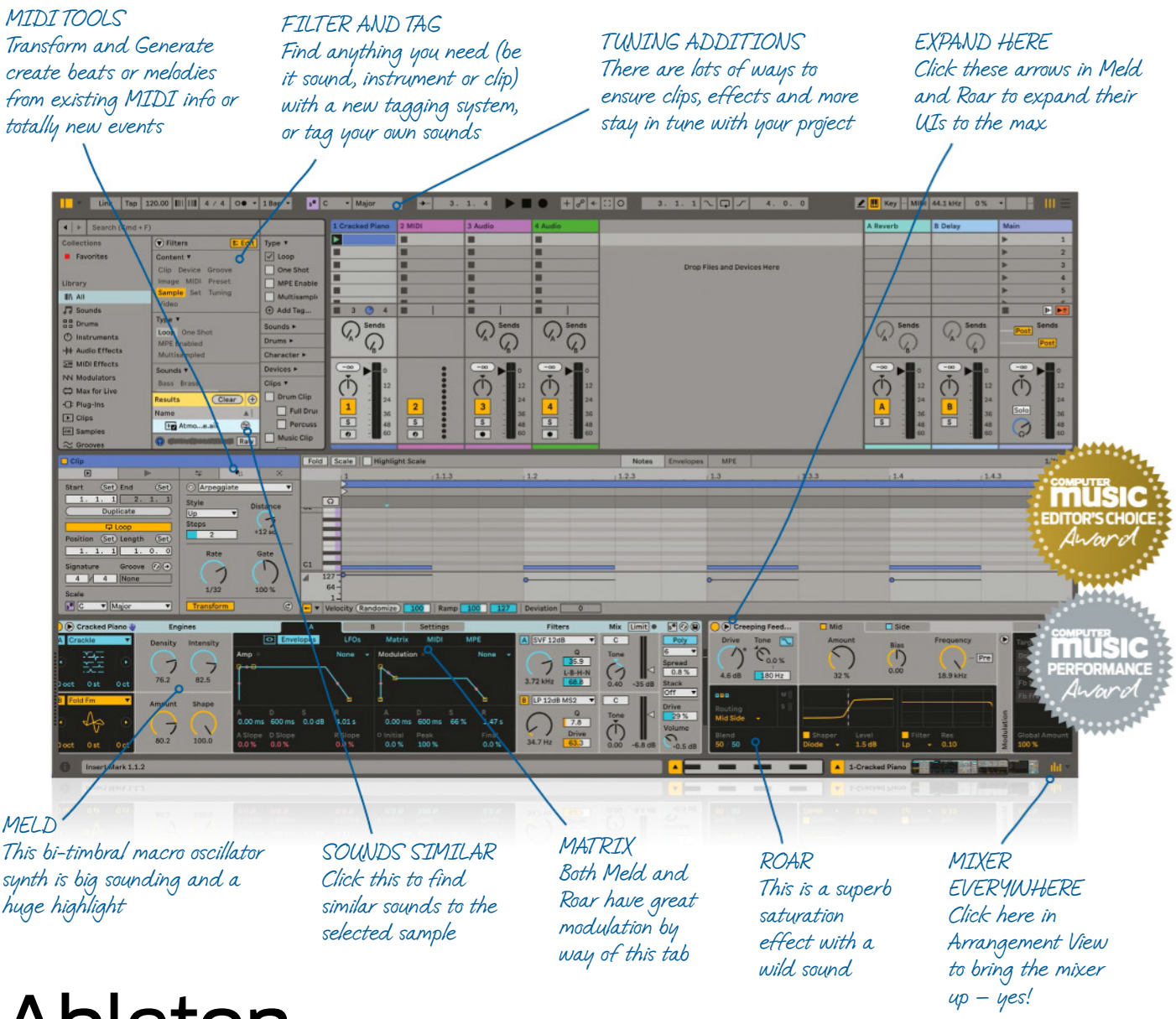
82 Soundware

Computer Music Innovation Award Awarded to products that challenge existing ideas and do something entirely new

Computer Music Performance Award A product has to really impress us with its functionality and features to win this one

Computer Music Value Award If the product exceeds expectations for its price, it will receive this gong

Computer Music Editor's Choice Award In the opinion of the Editor, the best product reviewed in the magazine this month



Ableton Live 12

£69/£259/£539 PC MAC

He's lived with it for a few months, now **Andy Jones** judges if Ableton Live's 12th major update is still the DAW MVP...

> You've got to love the way Ableton release big Live updates. It announces it and then spends months tweaking, opening up the beta for all sorts of feedback so when the final build is released, it should be damn near perfect. Which is great news for everyone, but it does feel like we've been talking about Ableton Live 12 for ages, and have certainly spent an inordinate amount of time with these various Beta builds. So, is the final (*final*) version worth your wonga?

What's all the fuss about?

Like many DAW users, we tend to get a bit too excited about big instrument additions when faced with a major point update, and largely go 'meh' to the smaller stuff, but Live 12 kind of turns that on its head. It has lots of both big and small extras, and in opposition to our normal upgrade use, we've actually been enjoying more of the less obvious stuff. But there's a lot to love overall, so let's try and give you an overview.

The big plugin headlines are the new Meld synth and Roar effect. Meld is a bi-timbral macro oscillator synth that has two selectable engines in which you can load very different oscillators, from basic VA to fairly obvious Bitgrunge to 'not immediately obvious' Tarp. Honestly the best first thing to do is load up one of the many excellent presets and just step through the oscillators; you'll soon get the Meld drift, and understand that it's hugely flexible because of

Roar has to be heard to be believed, and looks set to be an Ableton Live classic in the future

varied macro oscillators and their many combinations. But there's a lot more.

To the right of the oscillators, the controls will vary depending on which macro oscillator you have loaded in. Next to these controls are amplitude envelope, modulation envelope, and a couple of LFOs plus 17 filter options for each engine. So already you have a lot to play with within the bi-timbral setup, but the big theme of both Meld and Roar is modulation, and here's where it gets really good.

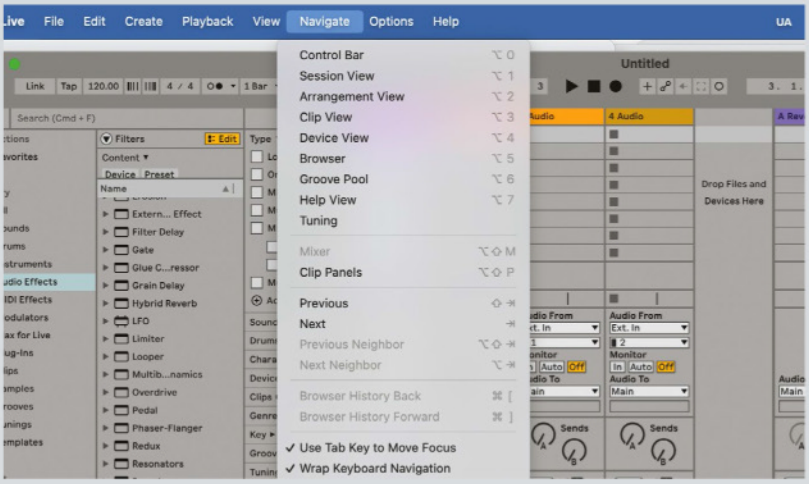
On a basic level, you can shape the two envelopes and LFOs directly, and a Matrix tab reveals them as simple sources which you can connect to common destinations. It's really easy to scroll up and down, click on a destination control and assign it to an envelope or LFO. But click on the top left arrow in Meld and you open up a much larger modulation matrix (see the bottom right pic) which opens up every control on the synth, adds MIDI and MPE sources, and even allows cross modulation between both macro oscillators. OK, it's not the prettiest way to take up more screen real estate, but it's one of the easiest modulation systems you could wish for, almost daring you to be more dynamic and creative with your synth sounds.

Roar deal

Meld, then, is a great way to step into the Live 12 update universe, but Roar will then take you by the hand and give you a good slap around the face. It's a saturation effect that can, should you wish, add some gentle colour to your sound. But it has three stages of saturation, various ways of combining them, massive modulation potential and an expanding UI like Meld. All this means that gentle saturation is the last thing you will use it for. This thing can add blistering dynamic effects, out-of-control wildness and gargantuan glitch, all of which makes it our Live 12 highlight. Like we said, we normally get excited about instruments, but Roar really is that good an effect.

Again, you should expand the UI to see Roar in its full glory, all three stages showing their individual saturation curves (choose from 12 shapes) and filter parameters. Where you played with the macro oscillators to get your head around Meld, here you play with the combinations of the three stages which can be Single, Serial, Parallel, Multi Band, Mid/Side and Feedback. Multi Band switches on all three stages on screen so is the most fun. Modulation is, again, easy to employ, although you only get four sources here (LFO 1 and 2, Envelope and Noise, which can be edited with their own tabs, or seen simultaneously when the UI is expanded) but can be applied to pretty much any multiple destination on Roar.

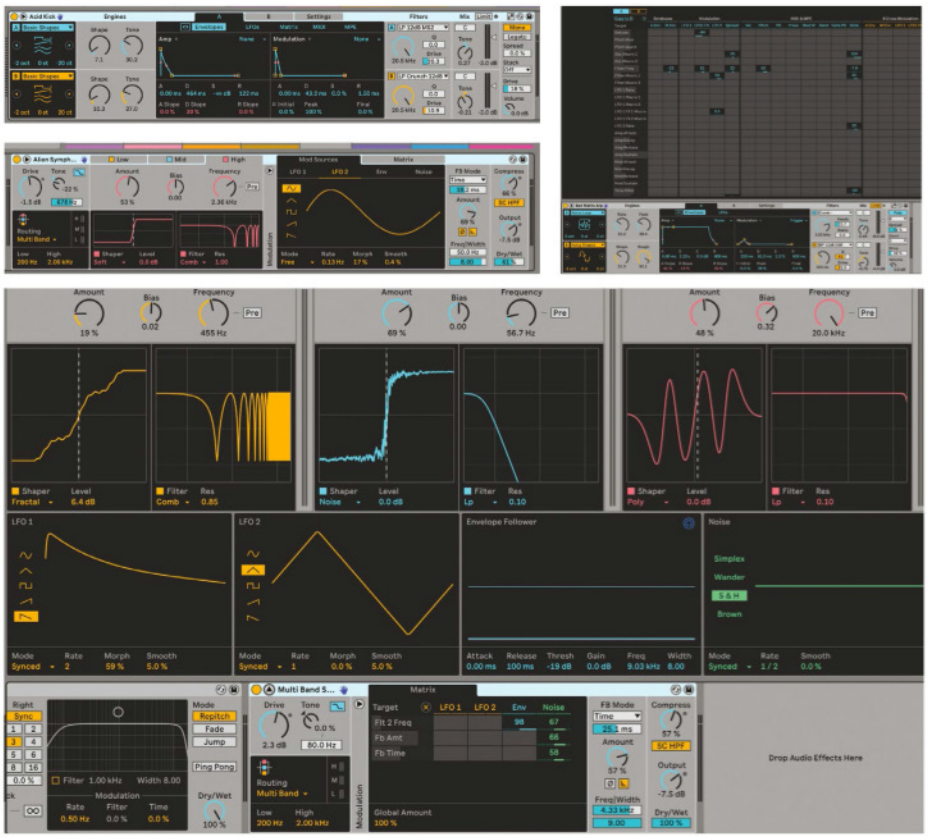
Roar has to be heard to be believed and also



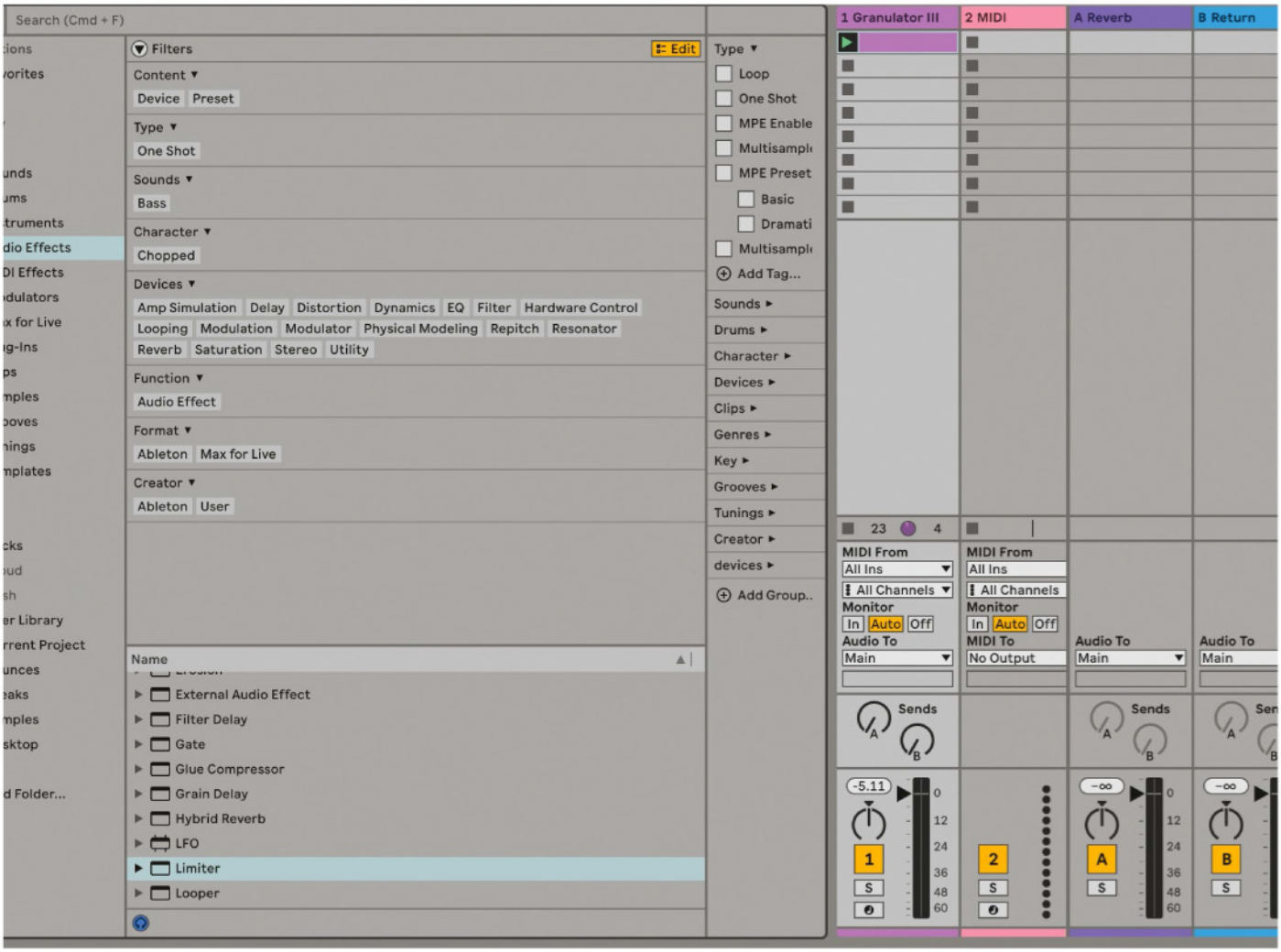
The new Navigate menu gives 1-click view access, and lets you Tab around each window with ease

Hidden gems

We'll try and round up some of the smaller but not necessarily less important updates here. Live 12 has improvements for visually-impaired users – the DAW works with braille displays and screen readers – and Ableton has also added lots of keyboard navigation extras for all users. The Navigate drop-down menu accesses various views by using the number keys, and when you select 'Use Tab Key To Move Focus', you can tab around the screen in your chosen view. Pressing Tab moves down to the next control and Shift Tab up to the previous, while Option Tab (or Command Tab in Windows) and Option Shift Tab move you right and left respectively on the same row. There are further options to refine these or use arrow keys, and other shortcuts have been updated. You can now solo with the computer MIDI keyboard enabled by shifting, for example, a previous annoyance now sorted. There's also a new Cmd>Opt>Shift>F shortcut for freezing tracks. All of these are welcome and are intuitive enough to become second nature if you're a keyboard whizz in Live.



The expanded (top right and bottom)and unexpanded (top left) UIs for both Roar and Meld. Keep them expanded!



The new filtering and tagging browser system helps you home in on sounds, or click the Edit button (in orange) and tag your own sounds and even create your own tag names

almost demands hands-on tweaking in the expanded UI. It is one of the most creative effects we've used, and looks set to be not just a Live 12 highlight, but a classic in years to come.

And talking of classics, a third Live 12 plugin extra comes by way of an update to Robert Henke's Granulator. This has always been an inspiring, leftfield tool, but now has a new UI, MPE support, and a capture feature to record any source directly in. If you are a fan of the kinds of granular mayhem from the previous version, you'll find this is more refined and flexible – a fabulous plugin.

Search me

Now remember what we said about the apparently not-so-exciting updates being some of the best in Live 12? Well, the new filtering and tagging system in the browser is just such a system. Combined with other ways to find sounds, samples and devices, it's a very neat addition, and one we didn't realise we really needed... until we started using it.

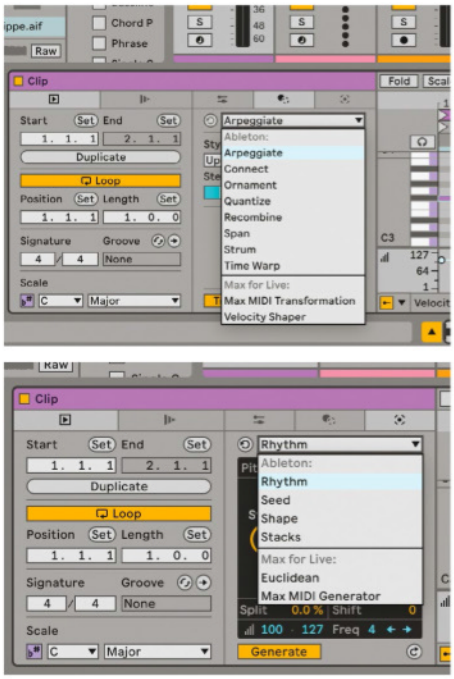
Yes, the new tagging system is a really slick way of letting you home in on anything – sample, MIDI file, groove, clip, whatever – by clicking options under categories like Sample Type, Instrument Sound, Character, Device, Format and a lot more.

The search system only works with Live content – it doesn't automatically tag your samples. But by clicking the orange Edit button, you open up more categories that you can use to tag your own sounds for future searches. You can also create your own tags to apply, and even save commonly used searches by hitting a plus button, and they get added to the browser.

One great additional feature is Sound Similarity Search. Here when you have a sample selected, you simply press a new icon on it and Live automatically finds similar sounds, with a bar ranking how close they are to the original. Sound Similarity Swapping also lets you swap sounds out in Simpler with similar ones.

Generating ideas

Next up, another big addition to Live in the form of Transformation and Generative MIDI tools, both new tabs in the Clip Title bar. The Generative ones create MIDI melodies and beats. Rhythm lets you create, say, kicks according to a number of custom-defined parameters. You can do this in real time, so as you adjust the three Steps, Pattern and Density controls, your kicks are generated automatically, or switch Generate off and hit a button to manually create different patterns each time. This not only works for beats, but



The new MIDI Tools are important additions with both Transformation (top) and Generative (bottom) types

Both sets of MIDI tools are great if you find yourself turning to the same progressions, beats and melodies – inspirational to use too

A 'why didn't they do this before?' addition lets you see both Clip View and Device View simultaneously

also with notes and melodies (using the Seed and Shape devices) and chords (with the Stack option) which can all be set to a particular key.

The Transformation tools let you change existing beats and melodies, with Time Warp, for example, allowing you to quickly create different snare rolls for example. Arpeggiate can split a chord into an arpeggiation, while Connect fills gaps between notes with similar chords and melodies, and Ornament adds frills such as flams. There are many more to try, and both sets of tools are great if you find yourself turning to the same progressions, beats and melodies, as we do, and inspirational stuff if you are found wanting creatively.

Amongst the remaining upgrades, these are worth a mention. The Scale Mode option is great so you can have an overall project scale and you can lock devices into it using their Use Current Scale option. There are similar pitch and time utilities in Clip view to transpose, fit to scale and more. Some surprisingly 'not done before' additions include being able to see the mixer in Arrangement view, and being able to see Clip View and Device View simultaneously. Like we say, small additions, maybe, but they make a big impact.

Living it up

While Live 12 might be grabbing you with Meld and Roar – and they can frankly grab us any day they like – it's the more 'behind the scenes' action that makes this one of the most complete Ableton updates in years.

The cost of upgrading to v12 varies depending on your existing level so login at Ableton to find out, but from v11 it's £179. Not cheap but we think there's enough here to warrant it, and if you are new to Ableton Live, then v12 is the most tempting proposition ever. It could well be the time to make the leap. **cm**

Web ableton.com
Pricing **Intro** £69 (16 x audio/MIDI tracks/scenes, 8 insts, 4 packs); **Standard** £259 (Unlimited audio/MIDI tracks/scenes, 13 insts, 16 packs, 12 MIDI tools); **Suite** £539 (20 insts, 33 packs, 14 MIDI tools, Max for Live). *Upgrade from v11: Standard £99, Suite £179*

Lost and Found

Performance Pack

Lost & Found and Performance are two new packs that comes with Live 12, too

Verdict

For One of the most well-rounded updates to Live ever
We love Meld...
But, we love Roar even more
The MIDI tools are inspiring and incredibly creative (and easy to use)
The new browser is intuitive and offers features we didn't realise we needed

Against Upgrade price is not cheap

With possibly more features across more areas than any other previous Live update, v12 is a triumph and solidifies the DAW's reputation as being one of the most creative music makers around

9/10

BLEASS Arpeggiator $39

PC MAC

Classic synth-based arpeggiators' up-down patterns can be tedious, says **Stuart Adams.** Can Arpeggiator bring the variability he craves?

> Arpeggiators have been a common feature of synths for decades, but many modern examples can blur the lines between arpeggiator and pattern sequencer. Features that increase randomness are fine for generative music and the like, but programming these advanced arpeggiators is typically a longwinded process that lacks the spontaneous playability of a classic – yet basic – synth arpeggiator.

Getting triggered

The new BLEASS Arpeggiator, developed in collaboration with French producer Canblaster (see our interview in issue 332), flips this state of affairs on its head, delivering complex arpeggio patterns via an intuitive, synth-like interface. To do this, the plugin breaks the arpeggiation process into a series of discrete stages.

The first of these, Trigger, creates the underlying clock that drives the arpeggiator. This can run freely or be synced to your DAW, with-or-without groove and time multipliers applied. More tantalisingly, the Trigger section can respond to an incoming MIDI note, allowing you to step through an arpeggio using a pattern, or manually by pressing a key.

The Trigger section feeds to the Pattern section. This defines 62 different rhythmic patterns of varying step lengths, with each defining the steps that will and will not sound. Here also is a Chance setting that adds a randomising element to the pattern output, and this operates even if the Pattern selection is set to "Off".

Rhythms and pitches

Each time the Trigger section sends a clock, the current pattern step is checked to see if it is active; if so, and if the Chance test is passed, a

> The process feels much more like you're playing an instrument than programming patterns

note will be generated. If no note is generated, then a Next/Mute setting determines whether Arpeggiator will move to the next position in the incoming note buffer (ie, the internal list of the notes you are playing), or will hold at the current buffer position.

What's important to remember here is that what comes next, in this context, depends on the mode of the arpeggiator – that is, whether it is using an up, down, up-down, or whatever mode. This is configured in the plugin's Sequence section, which provides a choice of 20 different modes, each determining the order in which the notes you play are used.

In addition, the Sequence section lets you specify note gate lengths, how many times a note will repeat before moving to the next note in the buffer, across how many octaves the sequence will repeat, and even the total number of times a sequence will play.

Where most arpeggiators will output only a single note at a time, BLEASS Arpeggiator is fully polyphonic, with each step triggering up to four notes, and the option to add a bass note to each step. Alternatively, with the Chord mode enabled, the plugin will play simultaneously all of the notes you are holding, applying the rhythmic pattern, gate lengths and transpositions determined by the Trigger and Pattern sections.

'Appy arps

Each instance of BLEASS Arpeggiator that you add to a project will create its own MIDI bus within your DAW, making it easy to send arpeggios to whatever synth you like (including external ones).

Alternatively, you can enable the plugin's built-in synth, a simple FM-based affair that's just right for creating the punchy, bright sounds that arpeggiate well. This is especially useful when it comes to the iOS app – just launch the app and start playing!

BLEASS Arpeggiator is quite unlike any other arpeggiator or pattern sequencer we've seen. Its synth-style controls make arpeggio creation an intuitive, creative, hands-on process that feels much more like playing an instrument than programming patterns. The price is a steal too, so be sure to grab the demo and give it a try. **cm**

Web bleass.com

BLEASS Arpeggiator's synth-like GUI deconstructs the arpeggiation process into a series of discrete stages

The plugin's synth-like credentials extend to the inclusion of modulators that interact with arpeggiator settings

Modulated arpeggios

BLEASS Arpeggiator's synth-like credentials go much deeper than the arpeggio generator itself – there are modulators here too! The plugin recognises incoming Mod Wheel messages, which can be mapped to up to two destinations, and includes a pair of X-Y pads. These can route their X and Y values to different parameters, and are particularly fun if using a touchscreen, although they can of course be assigned to external controllers too.

A pair of LFOs, lifted directly from BLEASS' Megalit synth, offer users the usual selection of waveforms, but these can be reshaped using smoothing, which controls the maximum rate of change of the LFO's value, and time shaping, which ramps the playback rate across each waveform cycle. Also lifted from Megalit is the Motion Sequencer, a step sequencer that outputs modulation values.

Finding good ways to use these classic modulators in an arpeggiator context takes some experimentation, but there are some great examples included in the factory presets. The only shortcoming is that the modulators can't target the built-in synth, and nor can their values be passed over to an external synth as MIDI CCs.

Alternatively

510k Seqund
€29
This innovative pattern sequencer has many probability-based easy polyrhythm features

Kirnu Cream
€35
Taking a step-sequencer-style approach, Cream lets you create compelling arpeggio patterns

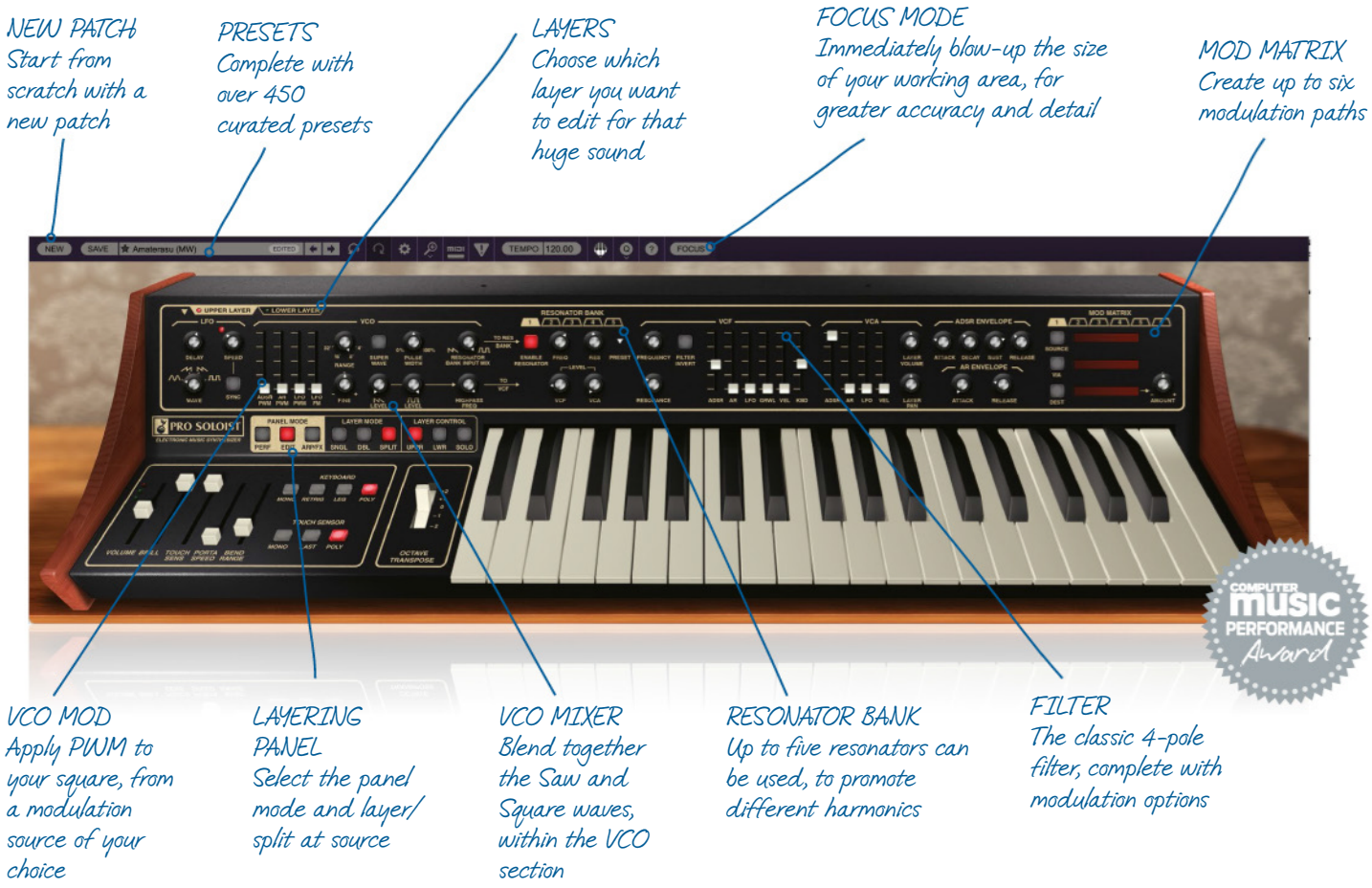
Verdict

For Easily creates complex arpeggios
Synthesiser-like modulators and performance controllers
Ideal for generative music

Against Modulators can't map to built in synth or send MIDI CCs

Successfully combines the intuitive playability of vintage arpeggiators with the complex patterns and harmonies of a modern pattern sequencer, then wraps the result in a hands-on synth-like interface

9/10



Cherry Audio
Pro Soloist $69 PC MAC

A bit of a diva in synth clothing? **Roland Schmidt** travels back to 1972 for a reimagining of an ARP classic with soloistic tendencies

> There was a time when the DAW du jour was a home organ, nestling in the corner of your lounge, waiting for Uncle Larry to spin a tune. But what would add a synthetic dimension to all those Bossanovas and Cha-Chas? A lead-line synth from Alan Robert Pearlman.

The ARP Pro Soloist came two years after its original sibling, the Soloist. Originally designed as a synthetic string, wind and brass solo section, to nestle on top of your home organ, the 'Pro' model refashioned the preset original to include editable control. It's therefore fitting that Cherry Audio has not only adopted the Pro Soloist for its latest software reincarnation project, but in true Cherry style, increased its potency exponentially.

The software Pro Soloist journey begins with a hearty collection of the original sounds, accessed through the Performance panel page. The quality of the analogue modelling is exemplary; the sense of nostalgia is heart-warming, providing a simple period-piece which could prove useful in certain settings. However, that's a mere entrée to the main event.

The legendary Soloist

Flick over to the Edit page and you'll find that you have immediate access to the nuts and bolts of each sound. Despite its humble origins, there's nothing simplistic about the output. We're in the subtractive domain, where the signal begins within the VCO section. Two volume pots labelled Saw and Square allow the blending of the initial signal, alongside additional features; the Saw can be multiplied into a SuperSaw, while the Square can be pulse-width modulated, either manually or from a LFO or Envelope source.

The signal is then uniquely routed to a high-pass filter, followed immediately by a Resonator Bank, where five sets of exceptionally rich resonators may be applied to colour the signal. It's at this stage that the Resonators input signals from the filter or amplifier sections, inducing feedback which injects some amazing colouration into proceedings.

Adding to this sonic character, the filter bank can create exceedingly throaty and organic filter

Armed with 450 presets, Pro Soloist has the capacity to impress with its diverse sounds and creativity

colours, and that's before it even reaches the main filter section, which supplies a beautifully modelled 4-pole ARP filter. There are numerous filter modulation sources, which include either envelope, LFO and velocity. In true ARP fashion, two envelopes are supplied; one full ADSR, and a more simplistic AR envelope, much like other ARP synths, such as the Odyssey or 2600.

Turn the page

If you had been hoping to take your modulation programming further, it's then possible to turn to the Modulation Matrix, which allows the comprehensive routing of every conceivable source to one of 58 destinations. You can also route via performance controls, if you're looking for momentary activation from a mod wheel or similar control element.

You also have the option to work in single, double (layered) or split modes. As you go on to access these latter modes, a lower panel will appear, duplicating the controls for a second round of programming, upon selection of the appropriate layer.

The third and final page exposes an equally comprehensive set of effects and an arpeggiator. Included effects range from phasers and flangers to distortion, delay and reverb. These all have an extensive armoury of control, to sculpt the back-end output.

More solo!

Cherry Audio has included the original synth's name and branding in this marvellous recreation, but the flipside is that that in itself may promote a small amount of confusion. The good news is that you can use Pro Soloist as ARP intended, as a monophonic machine, with acoustic similarities and tendencies, but you can also open up to full polyphonic operation, with broad-ranging editing, which is an unexpected and welcome surprise. Armed with 450 presets, Pro Soloist has the capacity to impress with its scope for diverse sounds and creativity. You might even say that it's actually less of a soloist and more of an almighty chorus! **cm**

Web cherryaudio.com

The original performance mode provides all of the glory of the original synthetic soundalikes

Banking the filters

We're all used to the common concepts of subtractive synthesis, where you take a harmonically rich source and send it through a filter, removing the harmonics that you don't want to hear. Sure enough, Cherry's Pro Soloist supplies both a beautifully wispy high-pass filter, along with an incredibly weighty 24dB low-pass filter. Even at this more simplistic end of business, the attention to the modelling of the filter weight is impressive, resulting in a very employable sound. However, the Resonator Bank adds another huge dimension, allowing the creation of up to five different harmonic characteristics, which then feed the low-pass filter. This is one of the secrets of the machine's design, as it allows the highlighting of certain harmonics, in order to help replicate the sounds of acoustic instruments. This was useful in a pre-sampling era, but in contemporary hands it's more of a creative component that can bring something unique to your sound.

The extensive effects page includes an arpeggiator which syncs seamlessly to your DAW

Alternatively

Sonic Couture The Attic 2
£179
18 characterful synths including curios reminiscent of the Pro Soloist (Kontakt player required)

Korg miniKORG 700S
$149
Korg's software version of the classic oozes charm; and you get a host of virtual effect pedals!

Verdict

For Surprisingly versatile with real hidden depths
The filter/resonator section is a complete treasure trove of sonic colour
Stunning effects section
Focus mode great for smaller screens

Against We're going to struggle here!

Cherry Audio are on a crusade, providing superbly different synths at attractive prices. An incredible gem; not to be missed

10/10

Synthogy Ivory 3 German D

$279 PC MAC

One of the most highly regarded piano libraries around gets an update. Does it strike a chord with our reviewer, Herr **Roland Schmidt**?

> Back in the early-to-mid '00s, sampling came of age. One of the companies at the forefront of the pursuit of multi-sampling perfection was Synthogy. Its original Ivory package was accompanied by rapturous applause from the professional and producer community, as it became an overnight industry standard.

Shuffle on 19 years and Ivory reaches its Mk3 phase, centred around an extensively sampled, Hamburg-manufactured, Steinway Model D-274. Steinway is arguably the most highly regarded piano manufacturer in the world, with the Model D considered to be the finest concert grand. It's certainly the largest, at 274cm in length, but with this size comes an authority in sound, which has established the brand as the primary choice for most concert halls.

It's therefore hardly a surprise that Synthogy has chosen to capture this European Model D by reuniting members of its established sampling team, in the prestigious venue of Le Domaine Forget de Charlevoix Concert Hall.

Under the lid

There are a considerable number of new features included in this sizeable upgrade. Firstly, Synthogy has a new sound engine called Real-time Gradient Blending (RGB). In layman's terms, it's an update to the real-time playback technology and DSP, taking the sound of the plugin to the next level. This translates to some impressive stats; Ivory 3 now responds to the MIDI 2.0 specification, although original MIDI controller users will still benefit from the usual 127 velocity points on the curve. This single piano will require 42GB of disk space, although Synthogy specify that this needs to be in SSD format, presumably because of the impacted speeds for reliable sample retrieval.

Impressive on so many levels, we simply don't have space or time to mention them all

Extensive features have also been added in the areas of sustain resonance, on-board mixing and multiple configuration of stereo microphone, all captured in the concert hall environment. There is also significant control available, throughout the plugin; the number of options for change in tone colour, via many of the instrument's components, means that while this piano undoubtedly has its own identity, it can be quickly and easily tweaked to yield the best results for the user.

Given the number of captured samples, instrument loading is not the fastest, but it does reward from the moment you start to play. The definition and authority that you hear, as you play up and down the full range of the instrument, is quite extraordinary. There's a slight unavoidable consequence, which translates as the bottom-end and lower-middle registers of the instrument sounding exceptionally strident. It does sound amazing, but these registers can sometimes overpower.

Ivory 3 includes a considerable number of presets, organised and curated by Synthogy, and scrolling through the various permutations delivers an impressive showcase. For transparency, we were employing some substantial full-range monitoring, which yielded significant bottom-end playback, which might also explain why we felt like we wanted to pull back the bottom end slightly, in favour of a clearer higher mid.

Final recital

Diving into the substantial parameter menus, we found it easy to subtly change the signal, altering its makeup to an output which always hits the mark. While auditioning the plugin, across classical, jazz and more contemporary settings, Ivory 3 did continuously reward. We particularly enjoyed being able to move the lid to various positions, including fully closed, which resulted in some excellent pianistic colours. Pianos are very personal instruments, and we loved the ability to tame your piano to suit your desired 88-note perception.

When we first started playing with Ivory 3, the sound that we were hearing didn't entirely resonate, and we have to question why? Out of the box, the tone is arguably darker and more tempered. Experiment with settings though, and you suddenly find a unique world of realism, which in piano terms makes this a Holy Grail. Ivory 3 is impressive on so many levels, we simply don't have space or time to mention them all. But have no doubt, all of the colours are there, and it's up to you to choose which of them best works for you. **cm**

Web synthogy.com

The extensive character and resonance settings give an extraordinary level of sound control over the instrument

Adapt the plugin and samples to suit your individual playing style, or controller velocity curve

Mic placements and instrument settings

Ivory 3 provides the user with an abundance of setting controls. The most accessible of these are the mic channels, which relate to the onboard mixer. Each channel adopts its own set of samples, so there is a loading and RAM implication, as each channel set is engaged. However, the options are extensive, from close microphone placements, to mid/side arrangements and a more distant ambient signal. This could go a fair way to alter the sound that you require, but the menu-bound character settings allow for change in timbre, key noise and hammer strength, to name just a few. You can also change the resonant response, and the nature of the Soundboard. Synthogy have additionally captured Una Corda samples, and you can even hear the identifiable Steinway sound of the dampers lifting from the strings, as you press the sustain pedal. There are also a number of additional sound sources, within the Synth section, which include a stunning set of acoustic String samples, supplied by AudioBros. If we said it was 'comprehensive', we would still be underselling the content.

Alternatively

Native Instruments Definitive Piano Collection
£179
This selection of three instruments provides a great collection of different piano timbres

Modartt Pianoteq
From $139
This modelled plugin suite offers various package sizes and prices, according to your preferences

Verdict

For Four layers of microphone capture, with onboard mixer
An amazing level of control
Enormously adaptable, both to your playing style and timbral output

Against There's only one piano included, but you can substantially change its character

A significant number of enhancements mean you can control just about every aspect of the instrument's majestic sound

9/10

Soundware round-up

Audiotent
Vapor £37

Super progressive studio-quality sounds for your consideration. They've all been plucked from a stack of iconic analog gear, and wrung through a processing chain so state-of-the-art, its patent for awesomeness is still pending.

Choose from 700+ loops and hits, painstakingly crafted to excite your track. There are basslines and synths to die for and drums that simply won't quit, plus folders of timeless textures, powerful pads, FX, chords and majestic music loops. It's the perfect pairing for all current, and as yet uncategorised, genres and sub-genres of dance music. Catch the vapours!

audiotent.com

9/10

Zenhiser
Peaktime £48

High-end techno samples, taking in a legion of loops, hits, MIDI files, and some staggeringly good full-track stems. This set weighs in at a beefy 5GB, and showcases nothing but serious sounds for kicking the party into gear.

Sweeping FX, devastating drums, bottomless basslines, and synths to incite a riot – it's all crammed in. All frequencies have been explored, and the full definition of 'techno' has been considered. If you're after a speaker-shaking session with some strictly main room beats, this adrenaline-spiking collection should pique your interest.

zenhiser.com

8/10

Bingoshakerz
Wild West Stories $24.95

Yee-haw! It's a rootin' tootin', boot-scoopin' set of 20 fully-loaded construction kits, slathered in Spaghetti Western fun and flavour. There's banjo plucking, fiddle diddling, drums, bass, slide gee-tars, and other dusty-bottomed favourites to rifle through.

Taking its cues from Sergio Leone and the grit of many a John Wayne flick, this 1.3GB haul of smoking samples is perfect for your next neighbourhood hoedown, or cowpokes on the lookout for cinematic and evocative samples.

bingoshakerz.com

8/10

Minimal Audio

Elevate $29

Cutting-edge vocals, fit for huge hooks, drops and more. The 280 phrases and variants in this energetic selection are cool, catchy, and sure to add a lift to any tracks left lacking. Everything is BPM and key-labelled for easy auditioning. And the recording quality is through the roof.

Seven talented vocalists stepped into the booth, so there's a solid range of delivery to choose from. Some dry, so you can get busy, and others run through the blender, if you want to leave the funky chopping to the glitch experts.

minimal.audio

8/10

ModeAudio

Bloom £20

A verdant crop of fully-grown guitar samples span this glorious garden of ambient delights. There are toppy tones, super low frequencies and other drones, riffs and strumming pulled from the strings. But, the picks of this 792MB collection showcase some mighty fine granular synthesis, with deliciously textured electric guitar and bass work in the mix.

The sounds inside range from 11 seconds to around a minute in length, so there's plenty of options. Blooming great!

modeaudio.com

7/10

Sound Ghost

Utopia £20

Drift away into a luscious land of languid loops. There are peaceful pads, dreamy drones and sun-kissed synths ready to whoosh all over you.

The 200 samples in this sumptuous set take in soothing and serene textures. Some from pianos and keys, others built from angelic vocals – all blissful, weightless, and expertly crafted to make your tracks soar.

The cinematic scope of the electronic sonics in here will find many an application in pretty much anything that requires layers of yummy and scrummy ambient tones.

soundghost.net

7/10

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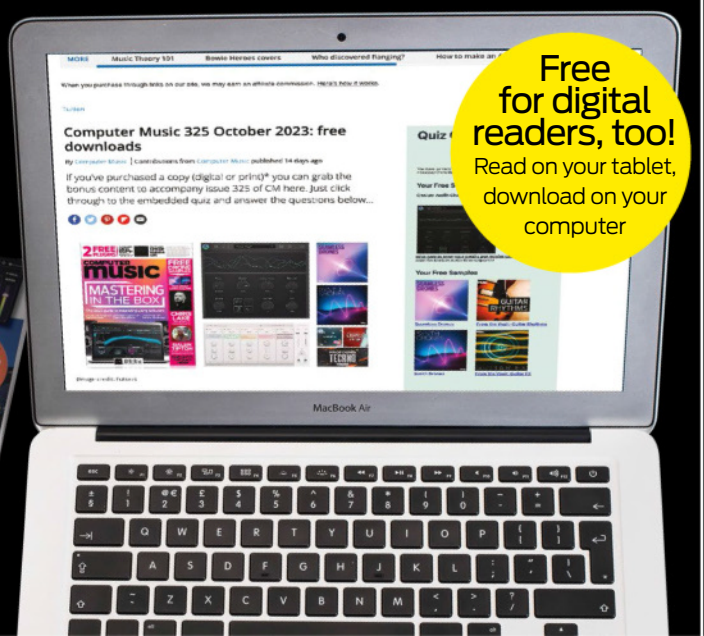

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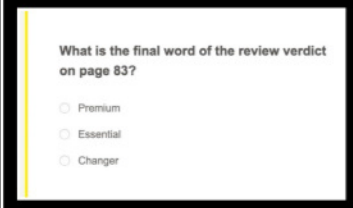

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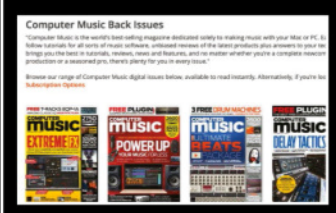

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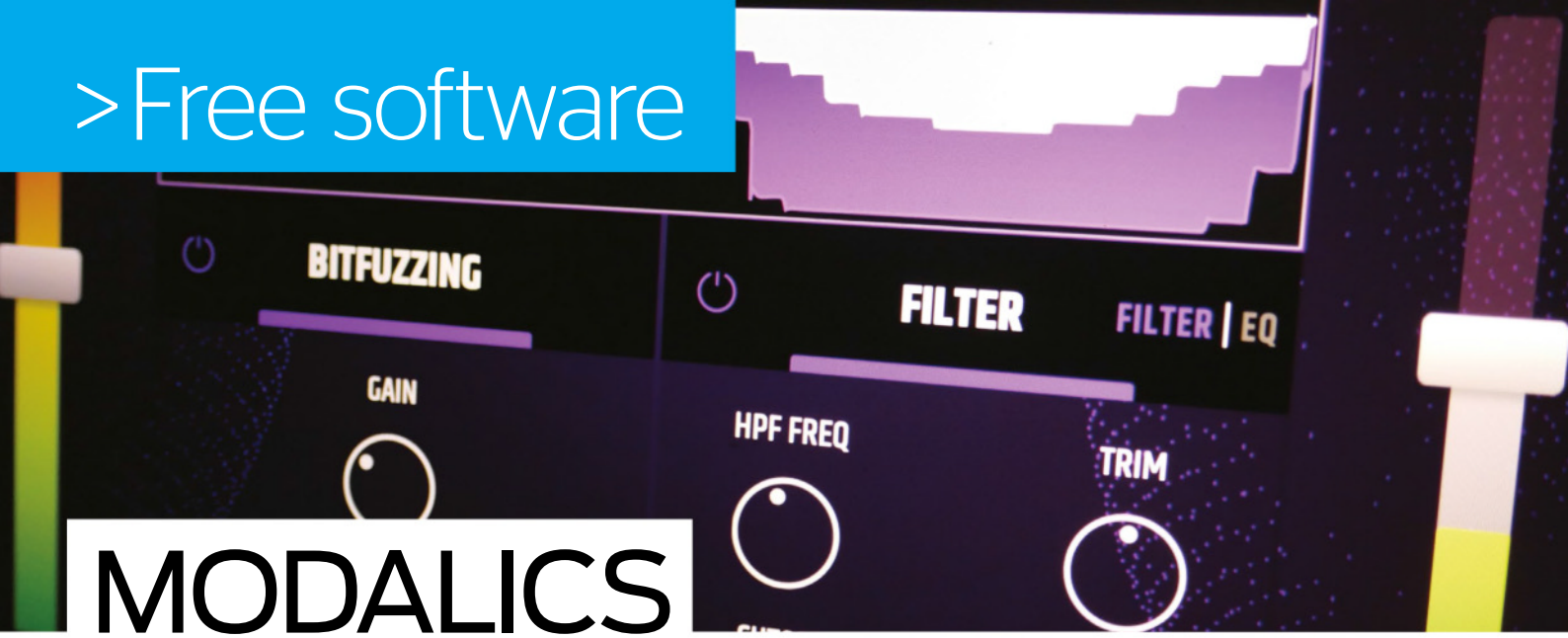


MODALICS BITFUZZER

PC MAC

Distortion can be a great effect for enlivening and mashing up sounds, and this month we've got you a plugin that can do both

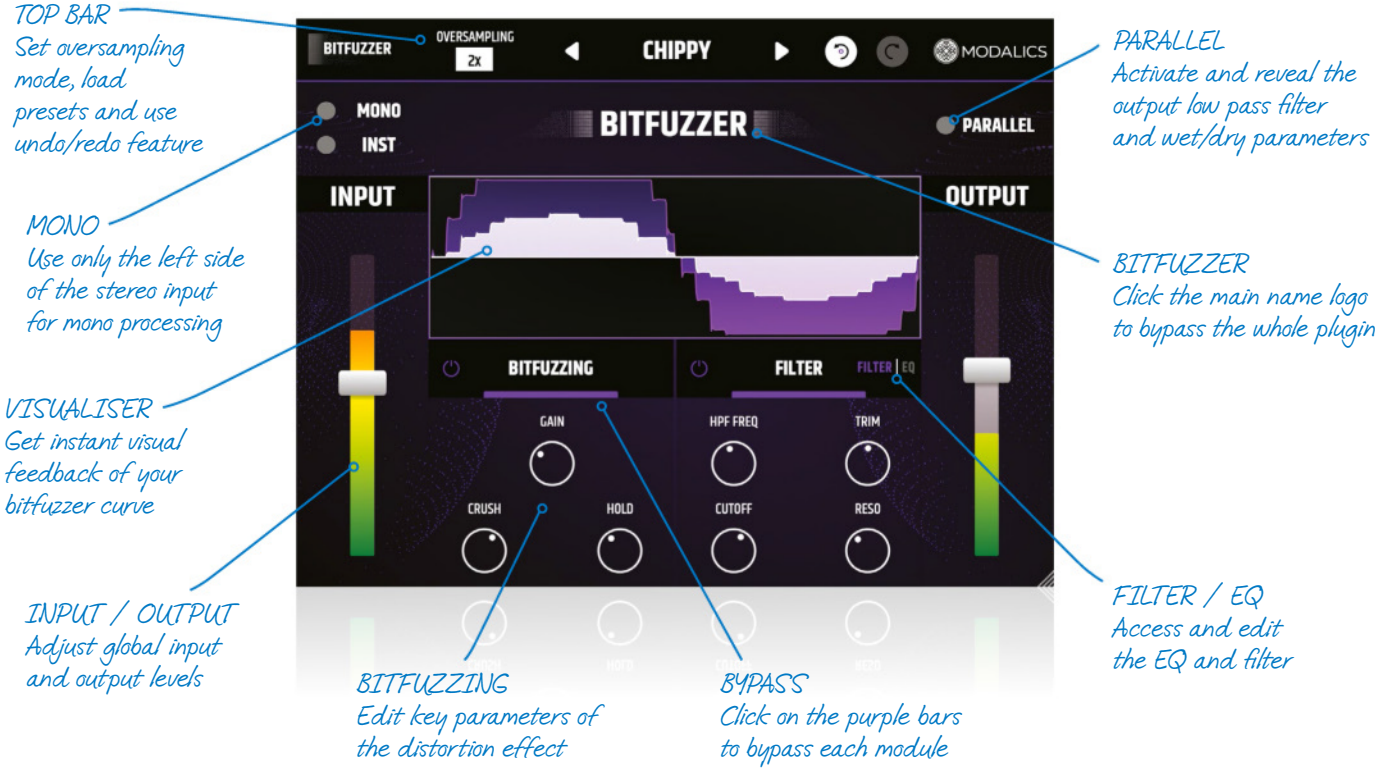
> Our last Modalics deal was a runaway success, so when the opportunity arose to feature another of their fantastic plugins, naturally we jumped at the chance. Said plugin is Modalics BitFuzzer, and this month this $29 plugin is yours absolutely free.

BitFuzzer (AU, AAX, VST, VST3) is a plugin with attitude. It combines bitcrusher, fuzz and distortion in what Modalics call an 'unholy alliance', and with presets such as Death Star and Disintigrate, who are we to argue.

The plugin features an easy to use interface with plenty of visual feedback, so you can see as well as hear the distortion curve applied to the signal. Sounds can be further finessed using the filter and EQ stages, while additional parameters help you tailor the outcome. Of course there are ample presets and these provide instant results or a good starting point for more extensive tweaking. There's also a handy parallel option to further expand the sonic flexibility.

In this tutorial we'll show you how to claim, register and download your free copy. Then we'll get you up and running, and take you on a tour of its key features. Then it's time for a more detailed look at what's included and what BitFuzzer can do, trying it out on some typical distortion targets. Along the way we should achieve Modalics' aim to 'destroy waveforms and make a beautiful mess'.

Finally, it's worth saying that Modalics have a growing stable of excellent plugins including the super creative Beat Scholar rhythm programmer, and their latest venture, MINDst Drums, which is a highly realistic acoustic drum kit instrument. If you want to know more about these and their other plugins then head over to www.modalics.com.



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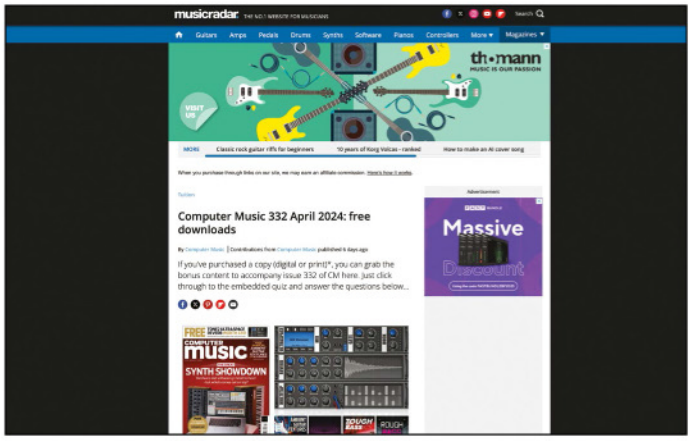
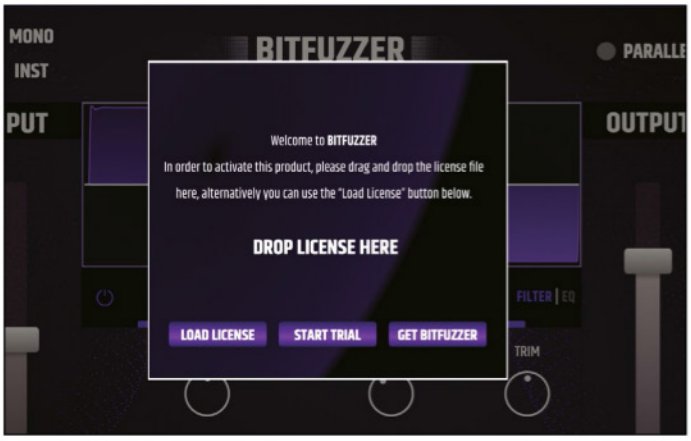
> Step by step 1. Installation





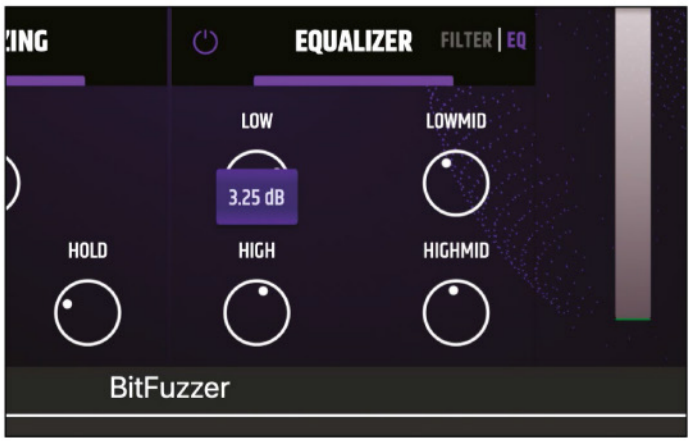
1 > To get started, head over to our content service and this month's page bit.ly/cm333downloads. With your issue to hand, answer the quiz questions and grab your BitFuzzer serial number. Then head to modalics.com/account/RedeemSerial and register or login, using your serial number to claim the plugin.

2 > Download both your installer and licence key, then run the installer. Launch your host DAW and load up an instance of BitFuzzer. The first time you do this a dialogue window will appear, and you can either browse to the licence key or simply drag and drop it on the dialogue. Click anywhere on the plugin and you should be all set.

> Step by step 2. Familiarisation

1 > Load up an instance and you'll see the interface follows a pretty standard layout with preset selection at the top. Then, working from left to right, we have the input options, the core controls in the middle (bitfuzzing, EQ and filter), and the output options on the right.

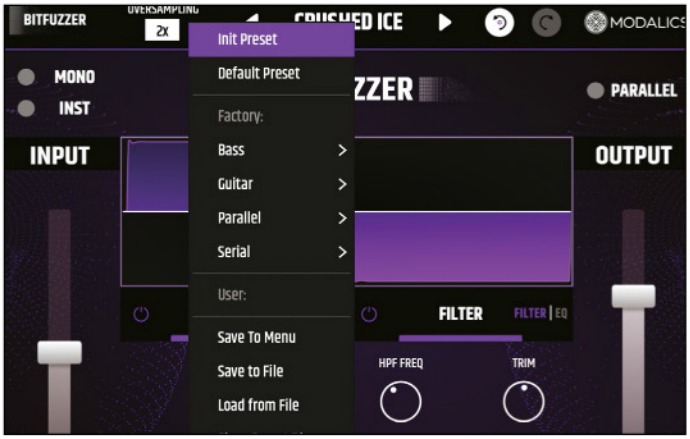
2 > This visualiser shows the effect of the Bitfuzzing section. Try adjusting the three bitfuzzing controls – Crush, Gain and Hold – and you'll see the curve changes shape accordingly. If you bypass this section, the curve reverts to a sine wave, meaning no distortion.



3 > BitFuzzer includes various ways to tailor the resulting distortion. The two-panel section to the right includes a high-pass filter, resonant low-pass filter and EQ. The EQ offers +/-20dB at four fixed frequencies.

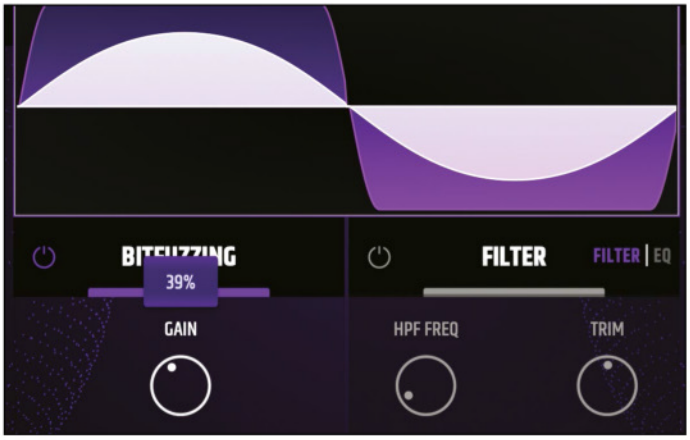
4 > The output section has two modes. The simpler default mode has a regular output level. Meanwhile, parallel mode reveals global wet and dry levels (-100dB to +6dB) and a resonant low-pass filter (22kHz to 50Hz). You can easily A/B the modes using the Parallel button.

> Step by step 3. Distortion 101

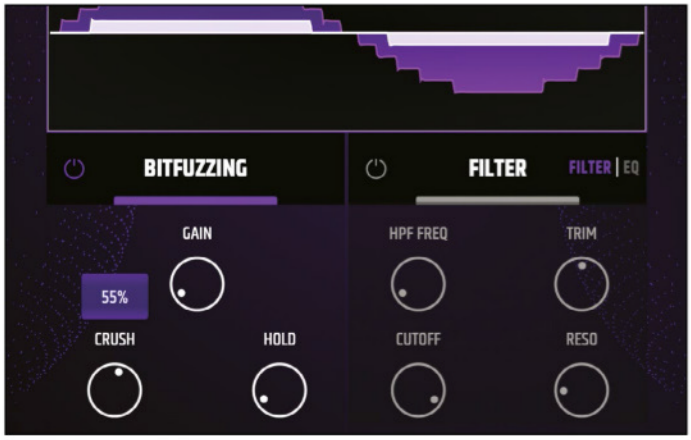


1 > When you load up the plugin it launches with the Crushed Ice preset, which is a pretty extreme example. To get a better idea of how the plugin modifies the input, we suggest starting with the Init preset. Click on the preset name in the top bar and select Init from the drop-down list.

2 > Deactivate the Filter/EQ section and make sure the Bitfuzzing module is active. Then load up a suitable audio source – we've chosen a beat loop. You'll see the default settings for bitfuzzing are all fully counterclockwise, and are thus set to their minimum values, and effectively bypassed.

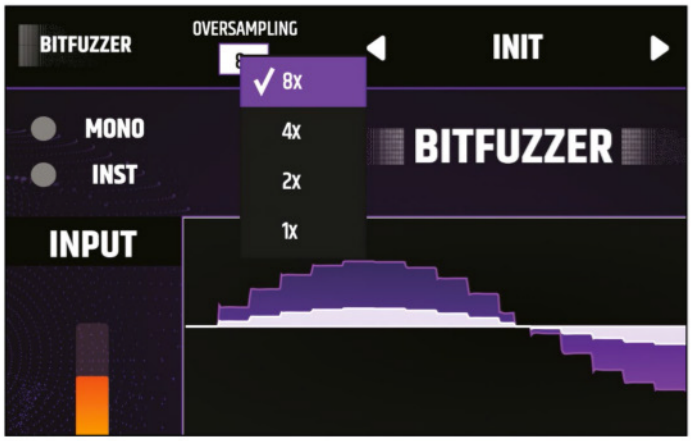


3 > Play your audio and you'll see a white realtime signal curve superimposed on the purple curve of the visualiser. If you now increase the Gain control the visualiser gradually changes to show a clipped signal. Sonically this achieves soft clipping initially and then as you keep increasing Gain you'll get harsher distortion and more harmonics.



4 > Now, reduce the Gain to zero and look at the Crush control. This primarily influences the bit resolution, and as we increase the level we hear more and more noise, and the signal begins to break up until at the maximum it's basically on or off. Spot how the visualiser gradually changes to show this.

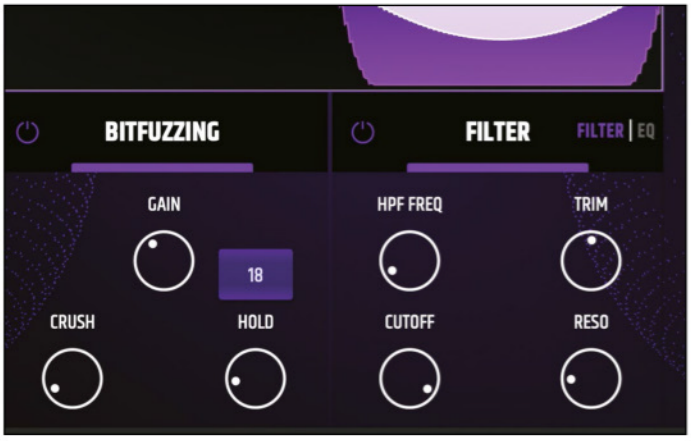
5 > Reduce the Crush setting to zero and now try the Hold control. This is a downsampling effect, and as you increase the control, a more ringy and noisy effect is created with the characteristic aliasing sound associated with sample rate reduction. Once again, check out how the visualiser modifies the curve.



6 > As mentioned, BitFuzzer includes oversampling, and this can help deliver a more accurate result. Sticking with the Crush option from the previous step, try some more accurate settings such as x4 and x8. You should hear a difference, and you'll also see some subtle changes to the visualiser.

> Step by step 4. Getting creative

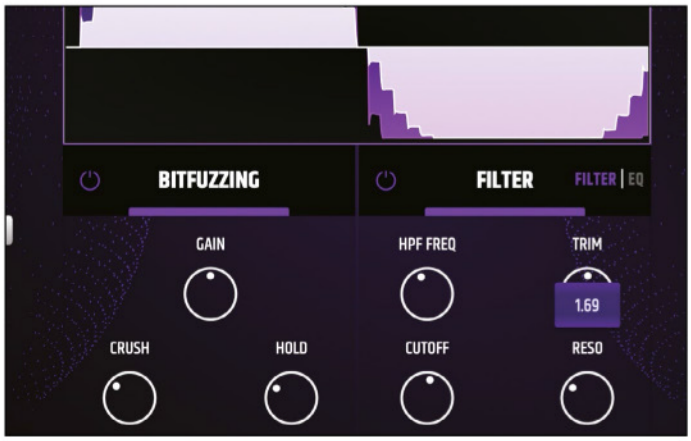
1 > BitFuzzer includes considerable additional shaping using the filters and EQ, and this allows us to get much more creative. Let's now work on a couple of sounds and see where this can take us. Load a suitable sound, we've chosen a simple bass synth sound that uses a couple of square wave oscillators and a low-pass filter.



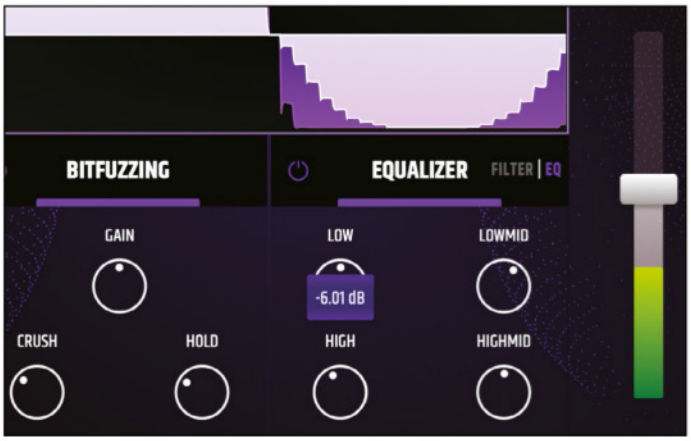
2 > If we now adjust the low-pass filter and resonance – we've set ours to about 4kHz and roughly a third (1.50) – we have quite an edgy and solid sound. To finish up, we've opened up the EQ panel, boosting the LowMid band by about 6dB, and to keep levels in check, lowered the overall Output by 6dB.

3 > Starting with the BitFuzzer Init preset and 8x oversampling, we've increased the Gain to about 35%. This adds harmonics and thickens out the sound, making it more solid and aggressive. We've then adjusted the Hold setting. Small changes here can make a big difference to the sound, and at about 18% we have a desirable ringing.

4 > Now let's try something more extreme. Returning to our beats loop we are aiming for a lo-fi degraded effect, which we can achieve by combining all three bitfuzzing processes with the filters and EQ. Starting with the Init preset, we've increased the Gain to about a half and then increased both Crush and Hold to about a third.

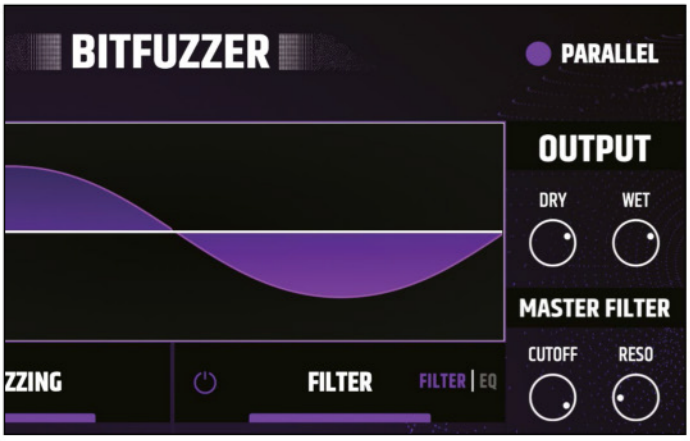


5 > The overall effect is too bright, so we can now shape this with the filters and EQ. First up, we've increased the high-pass filter (HPF) to about 100Hz to tighten up low frequencies. We then reduced the low-pass Cutoff to about 3.5kHz, increasing the resonance slightly to add some edge.



6 > The levels are now quite high, so we've reduced the main Output (about -6dB) and then opened the EQ panel to shape the overall sound. Here, a LowMid boost (about +6dB) and High cut (about -6dB) are enough to finesse the final sound.

> Step by step 5. Parallels

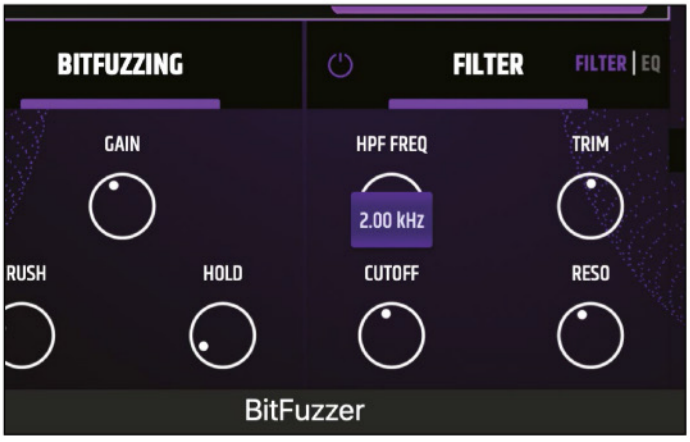


1 > As mentioned, BitFuzzer has two output modes. In the default state the plugin is essentially a serial processor, but select the Parallel option and you get independent wet and dry levels, and a global resonant low-pass filter. Parallel processing can be a great way to integrate more extreme distortion into your sound. Let's try this out.

2 > Drums and beats love a bit of distortion, but getting the right amount and flavour can be tough. Parallel processing can really help, and the classic 808 is the perfect candidate. Start with a completely plain 808 kick sound, and load up an instance of BitFuzzer, selecting the Init preset from the list.

3 > Now select the Parallel option in the output section, checking that the Dry level is set to minimum (-100dB) and the Wet level is set to 0dB. Also make sure that the master filter Cutoff is set to its maximum (22kHz). With these settings we can now shape our fully-processed wet sound.

4 > Start off in the Bitfuzzing section and adjust the Gain to taste (ours is at about 40%). You may also want to add in either or both of the other two effects – we've set the Crush to about 20%, which adds in a small amount of noise and some further artefacts, but have set Hold to zero.

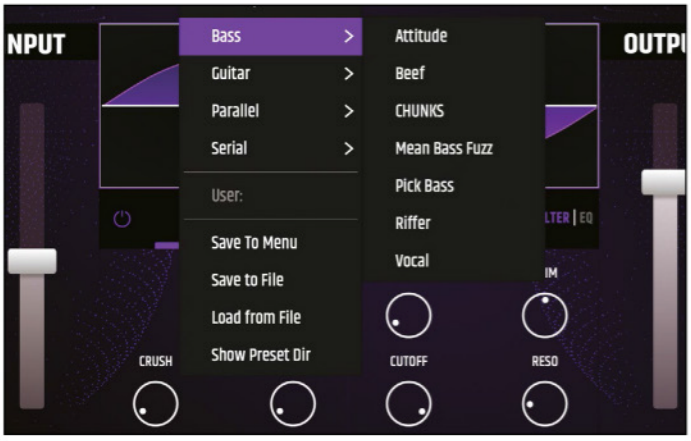


5 > To add some extra punch, try the filter section low-pass filter. We've set our cutoff to about 2kHz and then increased the resonance. This creates a further knocking boxy effect giving the 808 more low mids so it's more audible on small speakers. Even so, our kick now sounds pretty unusable.

6 > Now it's time to blend this sound with the original. Head back over to the output section and work on the balance between the Wet and Dry sounds. One good technique here is to set the Dry signal (ours is at about -3dB) and then gradually increase the Wet signal from its lowest until you have enough flavour.

> Step by step 6. Presets

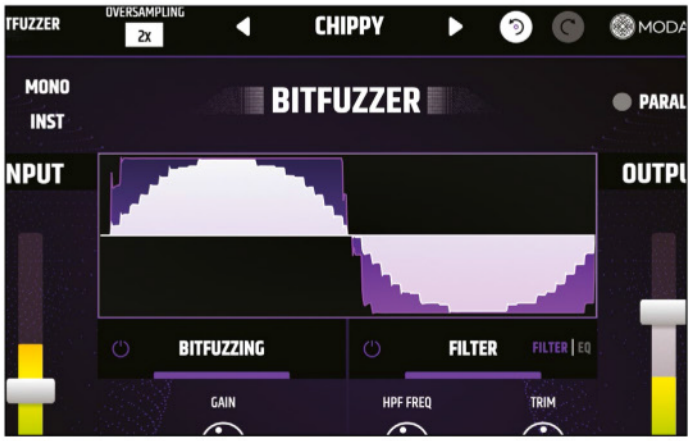
1 > So far we've created all of our BitFuzzer effects from scratch, but there will be plenty of times when you want a readymade starting point or simply want to surf for something new. BitFuzzer has a handful of useful presets categorised by instrument type or configuration. Let's take a look at what's included.



2 > If you click on the preset name, the drop-down menu reveals four categories – Bass, Guitar, Parallel and Serial. Naming in many cases is descriptive (Drive, Attitude, High Gain Lead for example), although there are some ambiguous options (such as CHUNKS). Many of the presets offer quite extreme distortion.

3 > You'll also find that, although there's a Parallel Processing folder, this mode features in many of the other presets. Bass, for example, is well catered for, and we particularly like the Beef and Pick Bass presets, both of which use the parallel mode coupled with some quite extreme distortion in the Bitfuzzing section.

4 > Sticking with Parallel Processing, there are only three presets in this task-specific folder. Filtered Together is an edgy bitcrushed sound, while Noise On Top is a very fuzzy sound. The most extreme would be Wet One's Beak, which features extreme degradation of the break-up variety.



5 > The presets are peppered with further extreme sounds that can completely reshape your audio. Sharp and Thin is much like a distorted gate, meanwhile Stay Strong sounds like a faulty circuit. Death Star delivers the classic ringy sample rate reduction, and Chippy does more of the same.

6 > Finally, there are no beat- or drum-specific presets, but BitFuzzer does have some presets that work really well after a bit of tweaking. A good start is Drive, but you'll need to tweak the cutoff and resonance of the output filter, increase the Dry mix and decrease the Wet mix to taste. **cm**

Download all assets for this issue at **bit.ly/cm333downloads**

Stutter, glitch and Buchla!

This month we're bringing you a selection of weird and wonky stutter and glitch sounds, as well as some slick West Coast Buchla synths

Stutter and Glitch

Based on some classic sampling techniques, Robbie's Stutter and Glitch samples add oodles of character

"When tremolo was not enough, the world of digital sampling brought a whole new sonic palette including the malfunctioning-on-purpose, chopped-up manglings that can be found in many, often unrelated, genres. From loping lo-fi repetitions to full on Aphex/Squarepusher slice 'n' dice, this sample pack makes use of a variety of stutter and glitching effects to produce a set of loops to adorn tracks, whether treated further, tucked behind beats or rammed right up front. Across 12 tempos these loops started life as acoustic drums, drum machines, monosynths, pads, acoustic/electronic pianos, and string sections. On slice 'n' dice duties were iZotope and BT's classic Stutter Edit, and the excellent ShaperBox 3 from Cableguys – this did the most work!"

More detailed info can be found in Robbie's PDF in the sample folder

SELECTED GEAR

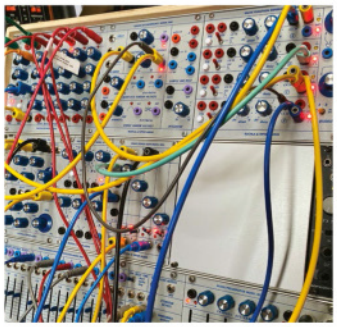
Arturia V Collection
NI Kontakt 7
Steinberg Nuendo 12
Boss SL-20 Slicer
Chase Bliss Mood
Focusrite ISA828 preamps
Eventide plugins
UAD plugins
Soundtoys plugins
Cableguys ShaperBox 3
iZotope Ozone 8 & Stutter Edit plugins
Unfiltered Audio Fault, Dent 2 and Sandman plugins
(+ more listed in Robbie's PDF)

West Coast Buchla

Patching up his impressive modular system, Oli has concocted a West Coast synthesis super-pack this month

"We've gone all West Coast this month using our Tiptop Audio Buchla Eurorack system. Lots of the loops are sequencer-based using the classic Buchla 5 steps, and often having the steps randomly ordered by the mighty 266t Source of Uncertainty module. Oscillator wise the 258t has two independent analogue oscillators that can be waveshaped and are designed to use lots of frequency modulation, which we did, often using each oscillator to FM its partner. We also used the sequencer running at audio rates as an unusual sound source for some of the patches instead of the oscillators. The 292t quad low pass gate and mighty 296t spectral processor were used in various ways to shape the sounds, from traditional LPG pings and Buchla bongos to experimental filter bank modulations."

More detailed info can be found in Oli's text doc in the sample folder



SELECTED GEAR

Tiptop Buchla 245t, 266t, 281t, 258t, 257t, 292t, 296t, 207t, Tiptop Audio stacked patch cables

FROM THE cm VAULT

Classic samples

Extreme packs

A few years back we went on a plunge into some of the most extreme sounds that we could spawn. Here are the results...

Cyclickstreme

The first of the extreme packs was crafted by Robbie at Cyclick Samples. Here he is to tell us more about it...

"Most genres of music have an extremist wing... well, except maybe Schlager, that is! The extremity most often takes the form of using the existing 'rules' and over-applying them, and in the case of much electronic, rock and pop music this over applying takes the form of turning everything up to 11 and overcooking sources (pardon the pun).

"Taking familiar instruments and treating them beyond the norms of audio production is a great way to explore the parameters of what's useful, like when, on occasion, chaining up a whole row of distortion pedals ends up producing something far more pleasing and beautiful than one would expect. With this in mind, the sample pack that's been created here uses extremes of production to mangle drums, guitars, pianos and organs into less easy-going-sounding instruments."

SELECTED GEAR

- **Focusrite ISA828 preamps**
- **Little Labs Redeye DI/ reamper**
- **Squier Telecaster Custom P-90**
- **Dreadbox Kinematic filter**
- **Keeley Fuzz Bender**
- **Dr No Octofuzz**
- **Fender The Pelt fuzz**
- **Sandpit PepBox clone**

Extreme Grooves

And here's Oli Bell's extremely extreme contribution to the Extreme Packs theme. Buckle up for some savage grooves...

"This set of samples is dedicated to more noise/glitch genres. The noise textures were made using layers of white and coloured noise from synths through effects. To add an extra level of variety, we used three favourite fuzz pedals: the Antichthon was self-patched using a splitter so that it fed back on itself and was then fed into the Guitar Disruptor and Rorschach.

"A vintage Realistic mixer was used with splitters and RCA connectors to patch all the mixer's somewhat limited outputs back into its own inputs (sometimes multiple times). By adjusting the faders we got all manner of tones and screeches.

"To finish, we hammered our favourite glitchy plugins to make some beats and loops. Alongside these we used our trusty Akai S01 that has been circuit-bent. The bends are accessible via a 5x5 switch matrix breakout box and add all sorts of grinding, glitching distortion and general mayhem to whatever you feed the sampler."

SELECTED GEAR

- **Circuit Bent Akai S01**
- **Realistic mixer**
- **Holy Island Audio Rorschach**
- **Electro Faustus guitar disruptor**
- **Catalinbread Antichthon**
- **Glitchmachines plugins**
- **Illformed Glitch 02**
- **iZotope Stutter Edit**
- **Various distortion plugins**
- **AudioThing Wires & Noises**

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Loopmasters

Loopmasters cm333 samples

This month's demo sample selection highlighting Loopmasters' latest releases

01 **Freaky Loops** Hard and Dark Drill
02 **Sample State** Deep Dub Techno House
03 **Element One** Ambient Techno
04 **Ghost Syndicate** Somnium
05 **Loopmasters** Kat Abel Nu Jazz & Neo Soul Vocals
06 **Industrial Strength** Toasted

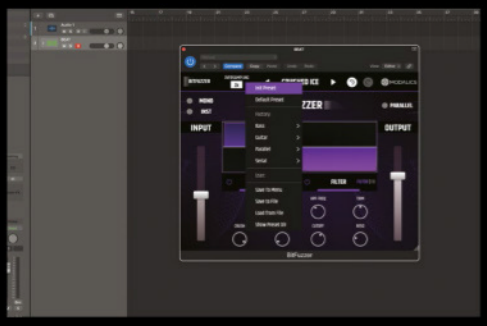
cm/video

Download this month's videos: **bit.ly/cm333downloads**

MODALICS BITFUZZER

See Modalics' fantastic bitcrushing, fuzz and distortion solution – yours free this issue! – in action

Read the full article from p88



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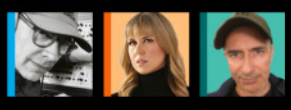

Our resident music production gurus walk you through their specialist field every month

CM TOOLS MASTERCLASS

BAZILLE CM FILTER KICK

Read the full article on p38

DANCE MUSIC MASTERCLASS

RAVE-STYLE TECHNO

Read the full article on p42

MIX MASTERCLASS

SIDECHAIN COMPRESSION

Read the full article on p46

Next issue

ON SALE WEDS 17 APRIL

Vocoders, prismizers, robots and rock!

Next issue we embrace our inner Daft Punk and take a deep dive into the use of vocoding, talkboxes and hard-harmonised processing to transform vocals. We'll highlight the best tools for the task and guide you through a range of voice-shaping tutorials!

PLUS!

Catching Flies
Gives us an insight into his software-based workflow

Free samples
Lay back and relax with the help of our chilled keys collection

On test
Steinberg Nuendo 13, GForce AXXESS and more...

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